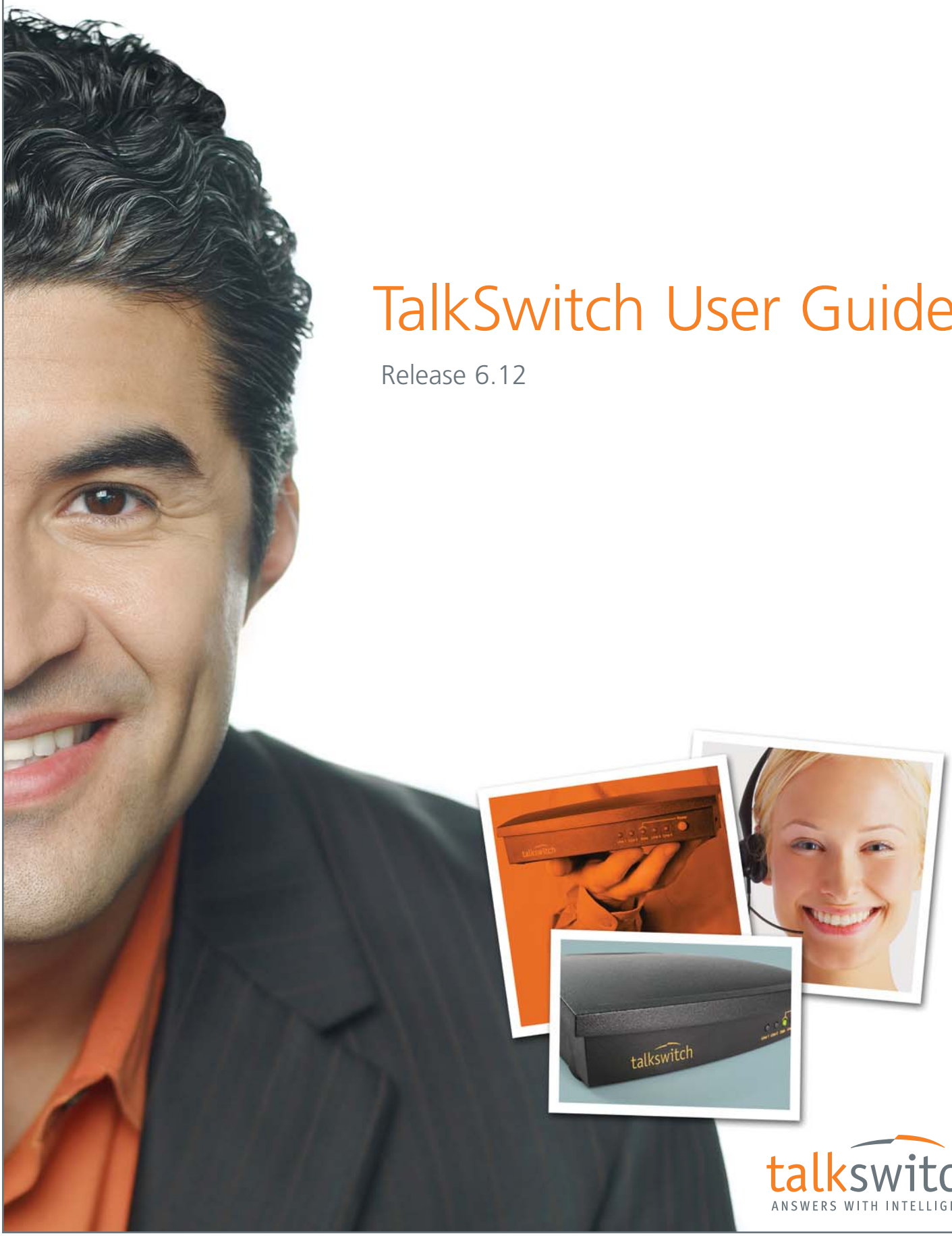
The **owner friendly™** phone system for small business



TalkSwitch User Guide

Release 6.12

talkswitch®
ANSWERS WITH INTELLIGENCE

Copyright Information

© 2009 TalkSwitch, a division of Centrepoint Technologies Inc. All rights reserved.

TalkSwitch®, the TalkSwitch logo, Concero®, answers with intelligence®, owner friendly™, seller friendly™, vendor friendly™ and *channel friendly™* are registered trademarks or trademarks of Centrepoint Technologies Inc. All other trademarks are the property of their respective owners.

Reproduction, adaptation or translation without prior written permission is prohibited, except as allowed under the copyright laws.

Information in this user guide is subject to change without notice and does not represent any commitment on the part of TalkSwitch. No part of this user guide may be reproduced or transmitted in any form or by any means, electronic or mechanical, including photocopying, recording, or information storage and retrieval systems, or translated to another language, for any purpose other than the licensee's personal use and, as specifically allowed in the licensing agreement, without the express written permission of TalkSwitch.

Release 6.12, October 2009
CT.TS005.003904

TABLE OF CONTENTS

PREFACE

TalkSwitch usage XIV
What's in this guide? XIV
What you should know XV
Single unit installation XV
Networked units installation XV
VoIP installation XV
Connecting devices XV
Important information XVI
Electrical shock XVI
Power and lightning surge protection XVI
Power adapter and power cord XVI
Power failure XVI
Change password frequently XVI
PBX fraud XVI
Emergency service numbers XVI
External audio source isolation XVII
Call redirection and service provider billing advisory XVII
Cleaning XVII
Finding the information you need XVII
Using the table of contents and the index XVII
Navigating with cross-references XVII
Where to go for further information XVII
Guide conventions XVIII

CHAPTER 1: TALKSWITCH INSTALLATION

TalkSwitch package contents 1
Recommended system specifications 1
Unit front panel 2
What the flashing lights mean 2
Line 1, 2, 3 and 4 lights flashing simultaneously 2
Power/Data and Line 1, 2, 3 and 4 lights flashing simultaneously 2
Line 1 and 4 lights, then Line 2 and 3 lights flashing alternately 2
Line 1 and 3 lights, then Line 2 and 4 lights flashing alternately 3
Line 2 and 3 lights flashing simultaneously 3
Line 1 and 4 lights flashing simultaneously 3
Power/Data and Line 1, 2, 3 and 4 lights on steady 3
Unit back panel 4
Installing the TalkSwitch management software 5
Installing the software 5
Upgrading the TalkSwitch software and firmware 6

Step 1 — Check current version 7
Step 2 — Download new software and firmware 7
Step 3 — Updating the firmware 8
Initial configuration 8
Select location for installation 9
Connect TalkSwitch to a network or a PC 10
Ethernet connection 10
USB connection 10
Internet connection 11
Connecting telephone lines 11
Connecting local extensions 11
Connecting a single-line analog phone 12
Connecting a two-line analog phone 12
Connecting an IP phone 13
Connecting a modem 13
Connecting a fax machine 14
Option 1 — Dedicated fax line 14
Option 2 — Distinctive ring 14
Option 3 — Automatic fax detection via the auto attendant 15
Connecting to the PA (public address) jack 15
Setting up music on hold 15
External audio source 15
Internal audio file 16
Networking TalkSwitch units on a LAN 16
Connecting TalkSwitch units to a LAN 16
Ethernet switch 16
Power up all the TalkSwitch units 17
Setting or changing the unit ID 17
How unit IDs affect system extension numbers 17
Keep track of the lines and extensions 18
Optimizing the system for networked use 18
Configuration settings 18
Outgoing line hunt groups 18
Auto attendants 18
Voicemail 18
Special considerations when connecting multiple units to a LAN 19
Increasing memory capacity 19
TalkSwitch memory cards 19
Installing a memory card 19
Upgrading TalkSwitch units 20
Enabling licensed add-ons 21
Verifying the connections 21
Connecting a TalkSwitch unit to a PBX 21

CHAPTER 2: TALKSWITCH CONFIGURATION

Introduction 22
Starting the TalkSwitch management software 22
About 25
Administration 26
Administration 27
Region selection 27
Language 28
Dial 9 or 0 routing 28
Scheduling 29
Modes 29

Change mode 30
Scheduling 31
System time 31
Date & time 32
Time zone 32
NTP server 32
IP Configuration 34
System IP settings 34
TalkSwitch proxy 35
Public IP address 35
Router port forwarding 36
Configuring the router manually 37
On-Hold/Ringback 38
On-hold settings 39
Transfer settings 40
Deleting a music on hold file 40
Playing music on hold on the PA system 40
Email Service 40
Email notification settings 41
Import an email list 43
E-mail server settings 43
Outgoing server authentication 44
Outgoing server port numbers 45
Incoming server authentication 45
Incoming server port numbers 45
Incoming mail server options 45
Testing the e-mail server settings 46
Managing voicemail messages 47
Setting up POP3 service with Microsoft Exchange 47
Risks of enabling POP3 service 47
Security recommendations 47
VoIP Configuration 48
Setting up a TalkSwitch profile 48
Activate profile 48
Provisioning details 49
TalkSwitch SIP server 49
TalkSwitch SIP client 49
Authentication 49
System VoIP options 49
VoIP Caller ID 49
Line reservation 50
Setting up a service provider profile 50
Activate profile 51
Automatic configuration 51
Manual configuration 52
Provisioning details 53
Viewing registrar entries 53
Viewing registration status 54
Setting codec options 55
Licensed Add-Ons 56
Local Extensions/Fax 57
Extension tab 57
Activate extension 58
IP extension details 59
TalkSwitch TS-550i, TS-480i, TS350i, TS-9133i or TS-9112i 60

TalkSwitch TS-850i 61
Polycom or Grandstream 61
Counterpath or other IP phone 61
Direct line access 61
Hotline access 63
Hunt group permissions 64
Caller ID settings 65
About call cascades 65
How call cascades work 65
Entering the call cascade 66
Being routed by the call cascade 66
Exiting the call cascade 66
Setting up call handling 67
Busy call cascade 67
No answer call cascade 69
Answered call cascade 70
Do not disturb cascade 72
Setting do not disturb mode 73
Ignoring call cascades for unanswered intercom calls 73
Ignoring call cascades for calls transferred from local extensions 74
Voicemail tab 75
Mailbox greeting 76
Caller options 76
Notification settings 77
Setting up notification by e-mail 77
Setting up notification by phone 78
Setting up notification by pager 79
Setting up the notification options 79
Setting up message waiting light activation 79
Voicemail screening 80
Adding IP phones 81
Overview 81
External IP extensions 82
Configuring the router 82
Configuring the router manually 84
Adding TalkSwitch TS-350i and TS-550i IP phones 85
Connecting the TalkSwitch IP phone to the network 85
Adding the extension to the TalkSwitch system 85
Activate extension area 85
IP extension details area 86
About programmable function keys 87
Programming TS-350i function keys 87
Saving a key assignment template 87
Using a key assignment template 88
Programming TS-550i function keys 88
Programming the TalkSwitchTS-350i or TS-550i IP phone as an external IP extension 89
If registration does not work 89
Adding TalkSwitch TS-9112i, TS-9133i and TS-480i IP phones 90
Connecting the TS-9112i, TS-9133i and TS-480i IP phones to the network 90
Adding the extension to the TalkSwitch system 91
Activate extension area 91
IP extension details area 92
About programmable function keys 93
Programming TS-9133i function keys 93
Programming TS-480i softkeys 94

Programming the TalkSwitch IP phone as an external IP extension 94
If registration does not work 95
Further configuration 95
Adding TalkSwitch TS-850i IP phones 96
Installing the base 96
Configuring the TS-850i as an IP extension 96
Configuring the TS-850i for use within the office 96
Activate extension area 96
IP extension details area 97
Programming the TalkSwitch TS-850i IP phone as an external IP extension 98
Using the TS-850i with a traditional phone line 98
Further configuration 98
Checking and updating firmware 99
TalkSwitch phone programmable key functions 100
Adding Polycom IP Phones 102
Adding the extension to the TalkSwitch system 102
Activate extension area 102
IP extension details area 102
Configuring the system for external IP extensions 103
Connecting the Polycom phone to the network 103
Confirming the Polycom firmware version 104
Programming the Polycom IP phone 104
Updating the Polycom firmware 104
Installing a TFTP application 105
Downloading and extracting firmware files 105
Configuring the phone to point to the firmware files 105
Adding Grandstream IP phones 106
Adding the extension to the TalkSwitch system 106
Activate extension area 106
IP extension details area 106
Configuring the system for external IP extensions 107
Connecting the Grandstream phone to the network 107
Confirming the Grandstream firmware version 108
Programming a Grandstream IP phone 108
Updating the Grandstream firmware 109
Installing a TFTP application 109
Downloading and extracting firmware files 109
Configuring the phone to point to the firmware files 109
Adding Counterpath IP phones 111
Adding the extension to the TalkSwitch system 111
Activate extension area 111
IP extension details area 111
Configuring the system for external IP extensions 112
Configuring a new SIP account 112
Updating eyeBeam software 114
Adding Other IP Phones 115
Remote Extensions 115
Extension tab 116
Activate extension 117
About call cascades 119
Setting up call handling 119
Voicemail tab 119
Mailbox greeting 119
Caller options 120
Notification settings 120

Remote extensions and telephone company services . . . 120
Ring Groups . . . 121
Activate ring group . . . 121
About call cascades . . . 122
Setting up call handling . . . 122
Ring group options . . . 122
General Voice Mailboxes . . . 123
Mailbox greeting . . . 123
Caller options . . . 123
Notification settings . . . 124
Voicemail Preferences . . . 124
Global message waiting indicator . . . 124
Mailbox setting . . . 124
Voicemail management . . . 125
View mailbox data . . . 125
Reset mailboxes . . . 125
Delete password . . . 126
Permissions . . . 127
Activate access codes . . . 127
Access code configuration . . . 127
Adding an access code . . . 128
Removing an access code . . . 128
Extensions requiring an access code . . . 128
Restricting extensions . . . 128
Clearing restrictions . . . 128
System Speed Dials . . . 129
Activate speed dial . . . 129
Creating a speed dial list . . . 130
Including the speed dial number . . . 130
Omitting the speed dial number . . . 130
Import system speed dial list . . . 131
Export system speed dial list . . . 131
Caller ID name tagging . . . 132
Telephone Lines . . . 133
Activate line . . . 133
Phone numbers . . . 134
Phone line services . . . 134
Line reversal settings . . . 135
Call handling . . . 136
Line optimization . . . 137
QUICK method . . . 138
OPTIMIZED method . . . 139
TalkSwitch Line Optimizer . . . 139
VoIP Numbers . . . 141
Activate VoIP number . . . 142
Phone number . . . 142
TalkSwitch profile . . . 142
Service provider profile . . . 142
Username and password . . . 143
Registration status . . . 143
Call handling . . . 143
Caller ID (or CLID) Based Routing . . . 145
Caller ID lookup list or CLID matching list . . . 146
Caller ID or CLID Routing Groups . . . 146
Setting up caller ID routing with an extension . . . 148

Line Hunt Groups . . . 149
Activate hunt group . . . 150
Hunt group line assignments . . . 150
Hunting order for outgoing calls . . . 151
Hunt group busy overflow for outgoing calls . . . 151
Overflow tone notification . . . 152
Auto Attendants (Menus) . . . 153
Activate auto attendant . . . 154
Actions during auto attendant playback . . . 154
Action performed after auto attendant playback . . . 156
Working with auto attendant messages . . . 156
Recording a new message . . . 156
Loading a message . . . 157
Erasing a message . . . 157
Example auto attendant . . . 158
Setting up the dial-by-name directory . . . 158
Setting the PBX extension length . . . 159
Auto Route Selection . . . 160
Introduction . . . 160
About automatic route selection . . . 160
About toll restriction . . . 160
About leading digits . . . 160
About carrier codes . . . 161
ARS and direct line access . . . 161
ARS and the 3-Way Calling/Conference service . . . 161
ARS and emergency service numbers . . . 161
Planning automatic route selection and toll restriction . . . 161
Setting up automatic route selection and toll restriction . . . 162
Activate automatic route selection . . . 162
Carrier code prefixes . . . 163
Emergency service numbers . . . 163
Auto route selection . . . 163
Restricting local extension access to specific hunt groups . . . 164
Allowing a local extension to bypass ARS and toll restriction . . . 164
Regulatory advisory notice . . . 165
Call Bridge (DISA) . . . 166
Using call bridge . . . 166
Activate account . . . 166
Hunt group access . . . 167
Auto Call Back . . . 167
Using auto call back . . . 168
Activate auto call back . . . 169
Account options . . . 169
Recording an announced message . . . 170
Prompted Call Back . . . 171
Using prompted call back . . . 171
Activate prompted call back . . . 172
Account options . . . 172
Recording an announced message . . . 173
File menu . . . 174
New Template . . . 174
Open › TalkSwitch Location . . . 175
Open › Configuration File . . . 175
Save . . . 176
Save To › File . . . 176
Save To › Location . . . 176

Retrieve Settings . . . 177
Close . . . 177
Exit . . . 177
Tools menu . . . 178
Memory Usage . . . 178
Voicemail Manager › Mailbox Status . . . 178
Voicemail Manager › Delete Mailbox Password . . . 178
Voicemail Manager › Reset Mailboxes . . . 178
Firmware Auto Update . . . 178
Posting firmware update files on your server . . . 179
Configuring TalkSwitch systems . . . 180
Setting up a TFTP server . . . 181
Setting up a syslog server . . . 181
Terminal Window (CLI) . . . 181
Syslog Output . . . 181
Call Logging Output (CDR) . . . 182
Defaults › Entire Configuration . . . 182
Defaults › Current Page . . . 182
Check for Updates . . . 182
Update Firmware . . . 182
Reboot TalkSwitch . . . 183
Options menu . . . 183
PA Output . . . 183
Transfer Preferences . . . 184
To local extension tab . . . 184
To remote extension tab . . . 184
To ring group tab . . . 186
From auto attendant tab . . . 186
From home phone tab . . . 187
Call Reminders . . . 187
VoIP Trunking . . . 188
SIP server registration . . . 188
VoIP ports . . . 188
Clock Preferences . . . 189
Internal Calls . . . 189
Call Back . . . 190
Set Date & Time . . . 190
Troubleshooting menu . . . 190
Auto Attendants . . . 190
Extensions › Regular Analog Extensions . . . 191
Extensions › IP Extensions . . . 192
Fax Detection . . . 193
Telephone Lines › Audio . . . 194
Telephone Lines › Detection and Timers . . . 198
VoIP . . . 200
VoIP ports . . . 200
SIP Server Registration Timer . . . 200
UPnP . . . 200
View Blocked IP Addresses . . . 200
Test Firewall . . . 200
Support . . . 201

CHAPTER 3: USING TALKSWITCH

Receiving calls . . . 202
With the auto attendant . . . 202

Without the auto attendant . . . 202
Using an analog extension . . . 203
Making calls . . . 203
Placing an intercom call . . . 203
Overhead paging . . . 203
Two-way paging . . . 203
One-way group paging . . . 203
Place out-of-office calls from a local extension . . . 204
Receiving calls . . . 204
For your local extension . . . 204
For any other local extension . . . 204
For a specific local extension . . . 204
While you're on a call . . . 204
Placing calls on hold . . . 205
Transferring calls to an extension . . . 205
Unscreened transfer . . . 205
Screened transfer . . . 205
Transferring calls to an outside number . . . 205
Unscreened transfer . . . 206
Screened transfer . . . 206
Parking and retrieving calls . . . 206
Parking a call . . . 206
Retrieving a parked call . . . 206
Queuing and retrieving callers . . . 206
Queuing callers at a local extension . . . 206
Queuing callers at a ring group . . . 207
Using call waiting . . . 207
Conference calling . . . 208
Two local TalkSwitch extensions and one outside caller . . . 208
Two outside callers and one local extension . . . 208
Using an IP extension . . . 209
Making calls . . . 209
Placing an intercom call . . . 209
Overhead paging . . . 209
Two-way paging . . . 209
One-way group paging . . . 209
Place out-of-office calls from a local extension . . . 210
Receiving calls . . . 210
For your local extension . . . 210
For any other local extension . . . 210
For a specific local extension . . . 210
While you're on a call . . . 211
Placing calls on hold . . . 211
Transferring calls to an extension . . . 211
Unscreened transfer . . . 211
Screened transfer . . . 211
Transferring calls to an outside number . . . 212
Unscreened transfer . . . 212
Screened transfer . . . 212
Call park — parking and retrieving callers . . . 212
Parking a caller . . . 212
Retrieving a parked call . . . 212
Queuing and retrieving callers . . . 212
Using call waiting . . . 213
Conference calling with TalkSwitch . . . 213

Two TalkSwitch local extensions and one outside caller . . . 213
Two outside callers and one local extension . . . 213
TalkSwitch phone programmable key functions . . . 214
Using VoIP . . . 215
Using a standard phone connected in parallel to TalkSwitch . . . 216
Making and receiving calls . . . 216
Transferring calls . . . 216
Modems and telephone line access . . . 217
Forwarding calls out of the office . . . 217
Automatic call forwarding . . . 217
Manual call forwarding . . . 217
Conditional call forwarding . . . 217
Transferring calls from a remote extension . . . 218
Screening options for forwarded calls . . . 218
Calls over VoIP with IP extensions and gateways . . . 219
Making calls . . . 219
Transferring calls . . . 219
Using the voicemail system . . . 219
Activating voice mailboxes . . . 219
Retrieving messages and accessing a voice mailbox . . . 220
Recording an announcement on a local extension . . . 221
Pager and cell phone notification . . . 221
Music on hold . . . 222
Music on hold and call forwarding to remote extensions . . . 222
Switching modes . . . 222
Switching modes manually . . . 222
Switching modes automatically . . . 223
Using call bridge and call back . . . 223
Using call bridge . . . 223
Using call back . . . 224
Activating call back . . . 224
Using prompted call back activation . . . 224
Using auto call back activation . . . 224
Accepting the call back . . . 225
Directly answering the call back . . . 225
Using the announced message option . . . 225

CHAPTER 4: CALL DETAIL RECORD LOGGING

Introduction . . . 226
Enabling call detail record logging . . . 226
TalkSwitch unit . . . 227
Email log file . . . 227
Retrieving call data records . . . 227
Call reporting . . . 227
Browser . . . 227
Analyzing the data . . . 228

CHAPTER 5: VOIP INFORMATION

About VoIP . . . 231
About SIP servers . . . 231
Selecting a location for the SIP server . . . 231
Connecting to a network . . . 231
Connecting to local and IP networks . . . 232
Connecting a VoIP gateway . . . 232

Confirming network capacity . . . 232
Configuring IP addresses . . . 232
Starting the TalkSwitch management software . . . 232
Setting the system IP settings . . . 233
Setting the public IP address . . . 234
Configuring the router . . . 235
Configuring the router manually . . . 237
External IP extensions . . . 237
TalkSwitch VoIP network . . . 239
Setting up a TalkSwitch profile for the SIP server . . . 240
Setting up a TalkSwitch profile for a SIP client . . . 241
Configuring VoIP numbers for a TalkSwitch VoIP network . . . 242
Setting up line hunt groups . . . 243
Advanced VoIP configuration . . . 243
Saving settings to TalkSwitch . . . 243
Verifying the TalkSwitch VoIP network . . . 243
Service provider VoIP network . . . 244
Setting up a service provider profile . . . 245
Setting codec options . . . 246
Configuring VoIP numbers for a service provider VoIP network . . . 247
Setting up line hunt groups . . . 248
Advanced VoIP configuration . . . 249
Setting up caller ID . . . 249
Reserving VoIP lines . . . 249
Saving settings to TalkSwitch . . . 250
Verifying registration . . . 250
FAQ . . . 250
How does VoIP work? . . . 250
Does a VoIP call sound like an analog phone call? . . . 250
What happens if the power goes out or if the IP network to VoIP fails? . . . 251
Can a firewall prevent VoIP calls from passing through? . . . 251
What is SIP? . . . 251
What is a VPN? . . . 251
Can a VPN help to carry data securely over the Internet? . . . 251
What's the difference between a public IP address and a private IP address? . . . 252
What is NAT and how does it affect VoIP? . . . 252
What's the difference between a static IP address and a dynamic IP address and what effect does this have on VoIP? . . . 252
What is a SIP server? . . . 253
What is dynamic DNS? . . . 253
What is the drawback of using dynamic DNS? . . . 253
How often will my public IP address change? . . . 253
What is a port number? . . . 254
TalkSwitch VoIP network administration form . . . 254

CHAPTER 6: TROUBLESHOOTING AND SUPPORT

Troubleshooting . . . 255
Configuration . . . 255
I am not able to retrieve settings from TalkSwitch . . . 255
I am unable to configure TalkSwitch using a touchtone phone . . . 255
Auto attendant . . . 256
The auto attendant does not play when calls come in . . . 256
The auto attendant message is broken up or very faint . . . 256
The auto attendant answers calls before any of the extensions ring . . . 256

The auto attendant is transferring calls to the wrong extension 256
The auto attendant answers calls, but does not transfer them to the extensions 256
Calls are going to my phone company voicemail instead of being answered by TalkSwitch 256
How do I access my voicemail remotely? 257
Music on hold 257
Callers hear the "One moment please" message and then total silence when the auto attendant transfers their call to an extension 257
Callers hear only silence when put on hold at an extension 257
Call routing 257
Callers are disconnected when transferring calls from one extension to another 257
I can't use my conference/3-way calling feature from the phone company 257
I'm unable to place intercom calls from a local extension 258
Answering and fax machines 258
The answering machine and/or fax machine picks up calls before they can be answered by telephone extensions 258
Incoming faxes are not automatically detected and routed to the fax machine. 258
Local extensions 258
I am unable to access TalkSwitch lines with a local extension 258
My PC (internal or external) modem cannot dial out 258
One of the local extensions (telephones) does not ring 258
Extension(s) ring, but there is no caller 259
When I try to access voicemail, I hear a busy tone. 259
Multiple TalkSwitch units. 259
I can't reach an extension across the LAN 259
The configuration does not show the other TalkSwitch unit(s) 259
After recording an auto attendant greeting, I can't play it back 259
Calls are not always reaching my voicemail 259
Calls across the LAN are lower in volume 260
VoIP. 260
I hear a re-order tone when I try to dial a VoIP location 260
Callers complain the sound is distorted or choppy 260
When I call someone or they call me, voice is only heard in one direction. 260
Support 261
To obtain warranty service 261
For products purchased in the U.S. or Canada 261
For products purchased outside of the U.S. and Canada 262

APPENDICES

Appendix A: Extension Numbers, Functions and Commands 263
Extension numbers 263
Functions you can enter from local extensions 263
Functions you can enter from outside phones 265
Functions you can enter in command mode. 265
Appendix B: TalkSwitch and Telephone Company Services. 267
Caller ID 267
Distinctive Ring 267
Voicemail 268
Call Waiting 268
3-Way Calling/Conference. 268
Transfer and Clear 268
Hunt/Rollover. 269
Other services 269
Services using call back 269
Services using prompted call back 269

Services and systems using Flash or Recall 269
Services using Ringback Tones 269
Services using line reversals 270
Appendix C: TalkSwitch and Power Interruptions 271
Appendix D: Safety and Regulatory Information 272
Appendix E: TalkSwitch One-Year Warranty and Return Policy 276
Appendix F: Specifications 278
Appendix G: Copyright and Licensing Notices 281
Appendix H: Home/Office Wiring Guide 284

GLOSSARY 291

INDEX 297

PREFACE

Thank you for purchasing the TalkSwitch phone system. Please read this section before you start installing the product.

TALKSWITCH USAGE

The TalkSwitch unit (part number CT.TS001.1) you have purchased is designed and supplied to meet the technical standards for use in specific countries and for connection to the major telephone service providers in those countries. It is subject to regulatory certification and compliances as detailed in the appendices.

TalkSwitch cannot provide technical support and warranty for models exported and used outside their intended markets. For availability of TalkSwitch models in other markets please visit **http://global.talkswitch.com**

WHAT'S IN THIS GUIDE?

The *TalkSwitch User Guide* contains all the information you need, whether you are installing a single TalkSwitch unit or multiple VoIP enabled units. It is intended to be a complete reference accompanying the *TalkSwitch Start Guide* that ships with every TalkSwitch unit.

This preface contains important information to help you minimize your installation effort and to get the most out of the features and the flexibility of your TalkSwitch system. We strive to make your experience with installation and configuration the most rewarding possible.

We would ask you to read important information concerning power interruptions and safety precautions contained in the appendices in order to ensure your TalkSwitch equipment is set up in the safest way possible while avoiding equipment damage.

WHAT YOU SHOULD KNOW

While TalkSwitch is customer installable, certain skills are required if you need to route cables or to configure a network. The following points will help you determine the required skills:

- Configuring the TalkSwitch system using the TalkSwitch management software can be performed by anyone with basic computer skills once the system is physically installed with proper networking equipment configurations (if two or more units are networked on a LAN).
- While most buildings are wired to accommodate TalkSwitch system setup, the need to route telephone and/or network cabling can occur occasionally. If your organization does not have someone with this skill set, we recommend the use of an outside telephony system technician.
- You may need to connect TalkSwitch to network equipment such as a router, switch or hub with a connection to the Internet, and to configure a firewall, computer and TalkSwitch for networked use internally and with Internet. If your organization does not have someone with this IT skill set, we recommend the use of an outside IT technician.

If you are installing a single, non-VoIP, TalkSwitch unit, anyone with basic computer skills can use the *TalkSwitch Start Guide* and this user guide to perform a full installation and configuration.

TalkSwitch system installations can be categorized into some general configurations, such as single unit installation, networked units installation, VoIP or non-VoIP installation, etc. This section helps you determine your best possible plan of action using this guide, according to your installation type. While not every possible installation scenario is detailed below, determine which one better fits your situation and use its guidelines as a starting point. *Chapter 6: Troubleshooting and Support* on page 255 provides solutions for common problems and the index at the back is a means to quickly navigate to specific areas.

Single unit installation

When installing a single, non-VoIP TalkSwitch unit, you can skip the sections marked with the Networked and VoIP icons.

Networked units installation

When installing networked TalkSwitch units, you require the sections marked with the Networked icon.

VoIP installation

When installing a VoIP-enabled unit, you require the sections marked with the VoIP icon. *Chapter 5: VoIP Information* on page 231 provides an additional resource.

Connecting devices

When connecting the computer, telephone lines, local extensions and other devices to your TalkSwitch unit(s), *Chapter 1: TalkSwitch Installation* on page 1 provides the information you need.

IMPORTANT INFORMATION

Electrical shock

Do not open the enclosure, except as instructed to install an upgrade kit. When installing an upgrade kit, disconnect the power adapter before removing the enclosure cover, and replace the cover before reconnecting the power adapter.

Do not expose the TalkSwitch to any liquids; that can pose a risk of electrical shock.

Power and lightning surge protection

Surge protection devices are recommended to protect TalkSwitch in areas subject to lightning or power surges. Please consult your reseller for recommended surge protection devices.

If TalkSwitch has been supplied with surge protection devices to meet local regulatory requirements, these devices must be installed as described in their installation instructions.

Do not connect or disconnect any telephone lines during thunderstorms.

Power adapter and power cord

Use only the power adapter and power cord supplied with your TalkSwitch. Check the supply voltage matches the rating on the power adapter label. If your supply voltage does not match the power adapter rating, please consult your reseller.

For safe operation, the TalkSwitch power adapter must be connected to a power supply socket with a third pin (ground /earth).

Power failure

The equipment will not operate when mains power fails.

In the event of a power failure, each TalkSwitch unit will connect extension E4 to phone Line 1 to permit calls (except-AU models). Calls must be dialed without a hunt group. All other TalkSwitch functions will not be available until power is restored. We recommend that a telephone that does not depend on mains power be available for emergency use.

Change password frequently

Setting and frequently changing the system password is recommended to avoid unauthorized changes to system configuration and settings.

PBX fraud

It is recommended that call bridge (DISA) and call back passwords are changed frequently to avoid unauthorized users making telephone calls through the TalkSwitch.

Emergency service numbers

Ensure emergency service numbers are not blocked by the toll restriction feature. Frequently used emergency service numbers are pre-programmed to avoid blocking, but other local numbers may be used in some locations.

Before routing any emergency service numbers to a VoIP service provider, check that they do handle emergency service calls and any conditions associated with this service.

External audio source isolation

We recommend that any external audio source be connected to the TalkSwitch unit using an approved Line Isolation Unit (e.g. A-Tick ( ) approved in AU).

Call redirection and service provider billing advisory

Use of the automatic route selection, toll restriction and call detail recording features does not imply any guarantee whatsoever by regulatory authorities, telephone service provider(s), TalkSwitch or its distributors and resellers, with regard to the accuracy of these features and that the use of such features may not be considered by a telephone company in any disputes which may arise regarding the accuracy of any subscriber's telephone account.

Cleaning

Use a slightly moistened cloth or an anti-static cloth to clean TalkSwitch. Do not use any solvents. Never use a dry cloth; electrostatic charges could damage the electronics in the system. Ensure, however, that no moisture penetrates into TalkSwitch, which can result in damage.

FINDING THE INFORMATION YOU NEED

The following user guide functionality helps you find the information you need quickly and enables you to skip the information you don't need. This will make your experience with this user guide and your TalkSwitch system installation and configuration much more efficient.

Using the table of contents and the index

The table of contents at the front of this user guide contains all of the section headings and page numbers throughout the manual. The index at the back of the user guide contains a keyword reference with page numbers. If you are using this guide online in PDF format, the headings and page numbers will contain hyperlinks, allowing you to quickly navigate to the sections you need.

Navigating with cross-references

Cross-references are contained throughout the user guide in order to help you access related information and illustrations quickly. As with the headings and keywords located in the table of contents and index, these cross-references contain hyperlinks, allowing you to quickly navigate to the sections you need.

Where to go for further information

The guides listed below and other guides can be found on the TalkSwitch software CD, in the TalkSwitch folder in the Windows Start menu once the software has been installed, and in the support section of our website at **http://global.talkswitch.com**.

- For basic installation and configuration instructions, refer to the *TalkSwitch Start Guide* that ships with every TalkSwitch unit.
- For information on setting up a TalkSwitch system to use VoIP, refer to the *VoIP Network Configuration Guide*.
- For instructions on configuring the TalkSwitch system to support an Internet Telephony Service Provider (ITSP), refer to the appropriate service configuration guide on our website.
- For instructions on installing and configuring IP phones, refer to *Adding IP Phones to TalkSwitch*.

GUIDE CONVENTIONS

The *TalkSwitch User Guide* uses the following text elements and icons as visual aids, making the manual more accessible.

Text elements	
Italic	Italicized text highlights labels within windows, references within the document, and references to other TalkSwitch documents.
Bold	Bolded text highlights window names, button names, menu commands, links, telephone keys, and text you are asked to type.
"Italic"	Italicized text in double quotes highlights TalkSwitch system voice prompts you hear from a telephone.

Icons	
	The Networked icon is used to mark sections with information pertaining to a setup where two or more TalkSwitch units are networked together on a LAN.
	The VoIP icon is used to mark sections with information pertaining to a setup where VoIP is being used.
	The IP extension icon is used to mark sections with information pertaining to the set-up and use of IP extensions.
	The Alert icon is used to mark sections that are important. They may contain warnings or cautions alerting the user to pay special attention.

CHAPTER 1

TALKSWITCH INSTALLATION

TALKSWITCH PACKAGE CONTENTS

TalkSwitch units (part number CT.TS001.1) ship with everything you need to get started. A power adapter, cables, management software CD and documentation comes with every unit. Refer to the *TalkSwitch Start Guide* included with the unit to verify you have everything you need. The package for VoIP-enabled units also contains a *VoIP Network Configuration Guide*.

If any of these items are missing, please contact your TalkSwitch dealer.

Caution: If TalkSwitch has been exposed to low temperatures prior to installation, wait until the system has reached room temperature before connecting the power cord to avoid condensation.

RECOMMENDED SYSTEM SPECIFICATIONS

- Windows Vista (32-bit), XP or 2000 operating system (fully updated)
- Though Windows 7, Vista 64-bit and Server 2003 are not officially supported, many TalkSwitch customers use these operating systems.
- 220 MB free hard disk space (only 100 MB necessary if Java Runtime Environment (JRE) is already available)
- 1 GHz CPU
- 1 GB RAM (Vista) or 512 MB RAM (XP or 2000)
- 1024 x 768 video resolution

Network equipment: If your system includes multiple TalkSwitch units, you will need an Ethernet switch. If you plan on using VoIP or configuring your system remotely, you will need an Ethernet switch (or router with an integrated switch) and a broadband Internet connection.

UNIT FRONT PANEL

The front panel consists of a power button, four line lights, and the power/data light.

Figure 1: Front panel

Light	State	Description
Line 1 to Line 4	On	Line is in use.
	Flickering	Line is ringing.
	Pulsing slowly	Line caller is on hold.
	Pulsing quickly	Line is engaged by a device that is sharing the line with TalkSwitch.
Power/Data	On	TalkSwitch is powered on.
	Flickering	The TalkSwitch unit is exchanging data with the computer via USB.
	Pulsing slowly	A voicemail message has been left in one of the mailboxes. This indicator is disabled by default, but can be enabled.

TalkSwitch models with eight telephone lines do not have lights for Lines 5 to 8.

What the flashing lights mean

The TalkSwitch unit flashes the lights on the front panel in different patterns to show diagnostic states.

Line 1, 2, 3 and 4 lights flashing simultaneously

The unit ID of this unit is in conflict with another unit on the same LAN.

This condition may occur for systems running Release 6.11 or earlier firmware. To resolve this, you can assign a different unit ID to each unit. For more information, see *Setting or changing the unit ID* on page 17.

Power/Data and Line 1, 2, 3 and 4 lights flashing simultaneously

The boot block is functioning, but the firmware needs to be updated or reloaded. Ensure you update the firmware using the latest version of TalkSwitch software available on the TalkSwitch website for your region. Note: the version posted on the website is recommended over the version on your installation CD.

1. Choose **Tools > Update Firmware**.

Line 1 and 4 lights, then Line 2 and 3 lights flashing alternately

The prompts are corrupted or not loaded. Ensure you update the firmware with the latest version. Do not use the version on the installation CD, because there might be a newer version.

1. Choose **Tools > Update Firmware**.

Line 1 and 3 lights, then Line 2 and 4 lights flashing alternately

The firmware and prompts have been updated, and the TalkSwitch unit needs to reboot.

1. Choose **Tools > Reboot TalkSwitch** or, on the front panel, turn the power button off and back on.

Line 2 and 3 lights flashing simultaneously

File system error. The stored greetings, music on hold, and voicemail messages may have exceeded the memory capacity of the TalkSwitch unit.

1. Choose **Tools > Memory Usage**. The **Memory Usage** window appears.
2. Check the *Time Usage Summary* area.
3. If the stored greetings, music on hold, and voicemail messages have exceeded the memory capacity, delete unnecessary files.
4. Choose **Tools > Reboot TalkSwitch**. The **Reboot System** window appears.
5. Select *Lock down system and reboot when all calls are complete.*
6. Click **Reboot**. The TalkSwitch system will reboot when all calls have been completed.

Line 1 and 4 lights flashing simultaneously

Voice mailbox error.

1. Choose **Tools > Reboot TalkSwitch** or, on the front panel, turn the power button off and back on.

Power/Data and Line 1, 2, 3 and 4 lights on steady

The boot block is corrupt.

1. Contact the Technical Support department at TalkSwitch.

UNIT BACK PANEL

Figure 2: TalkSwitch 240vs, 244vs and 248vs

Figure 3: TalkSwitch 280vs, 284vs and 288vs

Figure 4: TalkSwitch 480vs, 484vs and 488vs

Figure 5: TalkSwitch 840vs, 844vs and 848vs

The PF box between E4 and L1 represents power failure support. In the event of a power failure or loss of power to TalkSwitch, extension jack E4 is able to receive and make calls on Line 1. Note: not applicable in Australia.

Port	Purpose
MUSIC	The music connector is a 1/8" (3.5mm) mono phono jack used as an audio input for music on hold.
PA	The PA (public address) connector is a 1/8" (3.5mm) mono phono jack used as an audio output for overhead paging, voicemail screening and music on hold.
LAN	The LAN connector is an RJ-45 jack used for connection to a PC, connection of IP extensions, connection of VoIP lines, and unit-unit networking. There are three green LEDs on the right of the LAN port. The top LED indicates network synchronization (connection), the middle LED indicates data Rx (receiving data), and the bottom LED indicates data Tx (transmitting data).
E1–E8	The extension connectors are RJ-11 jacks for analog extensions.
L1–L8	The line connectors are RJ-11 jacks for telephones lines.
USB	The USB connector is a B jack for connecting to a PC.
MEMORY SLOT	The memory slot on the side of the enclosure is a socket used with TalkSwitch memory cards. Memory cards are not available in some markets.
POWER	The power connector is a receptacle rated at 16 VAC/2.0 A for connecting the supplied AC adapter input line.

INSTALLING THE TALKSWITCH MANAGEMENT SOFTWARE

Installing the software

The latest version of the TalkSwitch software can be downloaded from the support section at **http://global.talkswitch.com**.

1. If the TalkSwitch unit is connected to a USB cable, disconnect the cable.
2. Turn on your computer and insert the TalkSwitch CD into your CD drive. The install program starts automatically.
3. Click **NEXT** and follow the instructions.
4. If the TalkSwitch unit was connected to a USB cable, reconnect the cable.

   If you are upgrading from TalkSwitch configuration software 5.00, this concludes the installation procedure.

   If you are upgrading from TalkSwitch configuration software 4.10, the **Found new hardware** icon will appear, followed by the USB driver installation wizard.

   a) Proceed with the USB driver installation by clicking **Next**.

   b) Once the USB driver files have been installed, click **Finished**.

   c) Once the USB driver installation is complete, start the TalkSwitch management software.

   d) When the **Connection** window appears, select *USB*, and then click **Connect**. The Data LED on the TalkSwitch unit will blink when the connection is successful.

If you prefer to use Windows Explorer or if the install process does not launch automatically:

1. Double-click the **My Computer** icon.
2. Double-click the **CD-ROM/DVD** drive.
3. Double-click the **Startscreen.exe** icon and follow the instructions. After you click the **Finish** button, you will see the TalkSwitch icon on your desktop. This means you have successfully installed the TalkSwitch management software.

Upgrading the TalkSwitch software and firmware

We are continually looking for ways to enhance your communications capabilities. When new features are added, we provide our users with immediate access to an update directly from Windows **Start** menu.

1. Click **Start > Programs > TalkSwitch 6.12 > TalkSwitch Auto Update**.

   If your TalkSwitch management software is due for an update, a window informs you of the status of your software version. If you need an update, you have the option to download the update.

TalkSwitch Update Download
An update was found for your TalkSwitch software. Here are the details:
Installed version:
4.01.014
Available version:
4.10.025
README for available version:
TalkSwitch software copyright (c) 2007 Centrepoint Technologies Inc.
WARNING: This computer program is protected by copyright law and international treaties.
TalkSwitch Release 4.10.025 for the United States and Canada is now available for download.
Release 4.10.025 is a recommended update for all TalkSwitch 24-CA, 48-CA/CVA, and VS products, and is intended only for use in the United States and Canada.
WARNING: This software must not be used, and is not supported, in other markets. The application of this software upgrade to any TalkSwitch systems in use outside the United States and Canada, invalidates any warranty may result in a situation where the product's original software may not be recoverable.
Release Information:
Release: 4.10 Build: 025 Region: US and Canada Date: March 26,
Download
Cancel

2. Click the **Download** button to download the update. After the file is downloaded, the Auto Update program prompts you to install the update.

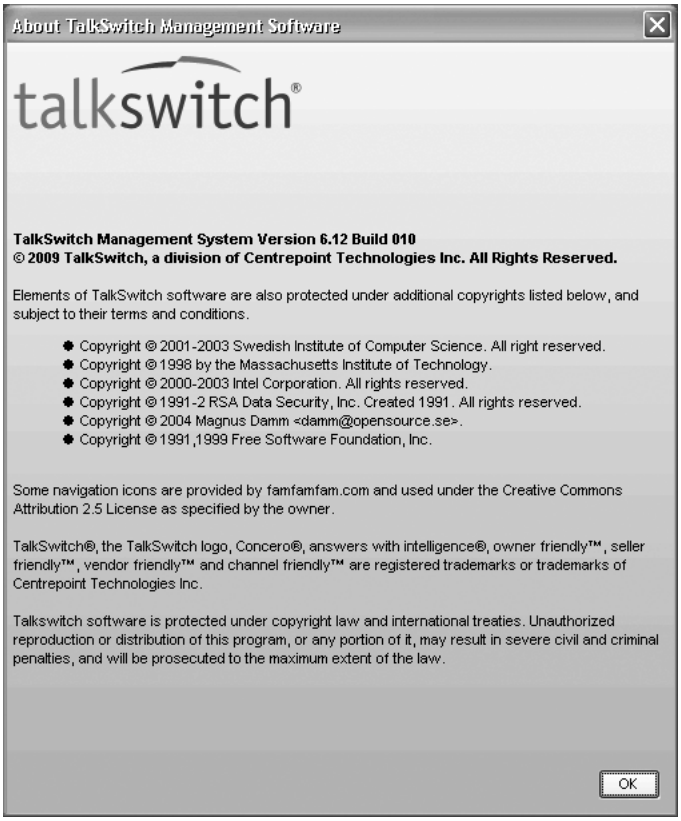
3. Click **Yes** to install the update. You can also install the update later from **C:\Program Files\TalkSwitch\TalkSwitch Configuration 6.12**.

The TalkSwitch management software must be closed in order to install the update.

Alternatively, you can check the versions you are running once the system is connected by following the three steps below.

Step 1 — Check current version

To identify the appropriate update file, you must determine your TalkSwitch software and firmware versions. To find your TalkSwitch software version number, choose **Help > About TalkSwitch Management Software**.


About TalkSwitch Management Software
talkswitch®
TalkSwitch Management System Version 6.12 Build 010
© 2009 TalkSwitch, a division of Centrepoint Technologies Inc. All Rights Reserved.
Elements of TalkSwitch software are also protected under additional copyrights listed below, and subject to their terms and conditions.
Copyright © 2001-2003 Swedish Institute of Computer Science. All right reserved.
Copyright © 1998 by the Massachusetts Institute of Technology.
Copyright © 2000-2003 Intel Corporation. All rights reserved.
Copyright © 1991-2 RSA Data Security, Inc. Created 1991. All rights reserved.
Copyright © 2004 Magnus Damm <damm@opensource.se>.
Copyright © 1991,1999 Free Software Foundation, Inc.
Some navigation icons are provided by famfamfam.com and used under the Creative Commons Attribution 2.5 License as specified by the owner.
TalkSwitch®, the TalkSwitch logo, Concero®, answers with intelligence®, owner friendly™, seller friendly™, vendor friendly™ and channel friendly™ are registered trademarks or trademarks of Centrepoint Technologies Inc.
Talkswitch software is protected under copyright law and international treaties. Unauthorized reproduction or distribution of this program, or any portion of it, may result in severe civil and criminal penalties, and will be prosecuted to the maximum extent of the law.
OK

Write down the software number. Go to the support section of the TalkSwitch website at **http://global.talkswitch.com**. The instructions help you select and download the appropriate upgrade.

The instructions on our website take precedence over any instructions in this user guide.

Step 2 — Download new software and firmware

When you double-click on the appropriate version for downloading, a window displays, asks you to enter a filename and select a folder for the download. We suggest leaving the filename as it is and saving it to your PC in **C:\Program Files\TalkSwitch\TalkSwitch Configuration 6.12**. Once the download is complete, begin your upgrade by double-clicking on the install file or using the **Run** command on the Windows **Start** menu. A series of windows will guide you through the installation process.

This process updates your TalkSwitch management software to a newer version. You still need to update the firmware to take advantage of new features. The new firmware file is placed in the TalkSwitch directory after running the install program.

Step 3 — Updating the firmware

The last step is to update the firmware. Make sure your TalkSwitch is connected to your PC.

1. Open the TalkSwitch management software.
2. Choose **Tools > Update Firmware**.

The update time varies depending on the connection type, how many files are needed and how many units are updated. On average the process takes between 1 and 5 minutes. The LED lights on the TalkSwitch front panel show diagnostic indicators for the update. See *What the flashing lights mean* on page 2. When the update is completed, you are prompted to reboot TalkSwitch. In the window, click the **Proceed** button.

Should the firmware update fail, please repeat the update process above. If the problem persists, contact your reseller or TalkSwitch.

Most firmware updates do not alter your system configuration. In certain circumstances, during a major system update, the configuration file may get reset to default values.

When you configure a unit for the first time and every time you modify the configuration, we recommend you save your configuration file to your PC and preferably to removable media (3.5" Floppy, CD-ROM disc or Flash stick) as a back-up. This avoids having to reconfigure the entire system in case of a hardware failure.

If you have to replace your computer, please download the latest install file from our website and follow steps 2 and 3 above. Run the install program. In case of a major system update, copy the back-up of your configuration file to
C:\Program Files\TalkSwitch\TalkSwitch Configuration 6.12.

INITIAL CONFIGURATION

Once your software is installed and your TalkSwitch is connected directly to a PC or via a switch/hub to the LAN, you can customize the TalkSwitch system with the TalkSwitch management software.

If you connect TalkSwitch to your telephone lines, it will do the following:

- An incoming call on any line will ring all local extensions.
- A receptionist can manually transfer calls to local extensions or voicemail.
- Every local extension will have a voice mailbox.

Recommendation: To minimize the disruption time of your telephone lines during the system setup, we recommend you configure the unit before you connect all the local extensions and telephone lines to the TalkSwitch unit.

SELECT LOCATION FOR INSTALLATION

Select a suitable location for installation of TalkSwitch.

TalkSwitch can be installed on a desktop or mounted on a wall. It can be installed in a office or communications or utility room.

Choose a location that meets the following requirements:

- Ambient temperature between 0–35°C.
- Away from heat sources and direct sun light.
- Avoid rooms with excessive humidity or dust.
- Within 1.5m of a power socket.
- Close to telephone company demarcation point for telephone lines, or sockets, and any in-building telephone wiring connection point.
- Away from any electronic devices that emit radio signals.
- Away from hazardous or explosive materials.

When installing multiple TalkSwitch units, do not stack units on top of each other.

If TalkSwitch has been exposed to low temperatures prior to installation, wait until the system has reached room temperature before connecting the power cord to avoid damage due to condensation.

Caution: Use only the power adapter and power cord supplied with your TalkSwitch. Check the supply voltage matches the rating on the power adapter label. If your supply voltage does not match the power adapter rating, please consult your reseller. For safe operation, the TalkSwitch power adapter must be connected to a power supply socket with a third pin (ground/earth).

Caution: Surge protection devices are recommended to protect TalkSwitch in areas subject to lightning or power surges. Please consult your reseller for recommended surge protection devices.

CONNECT TALKSWITCH TO A NETWORK OR A PC

Connect the AC adapter provided with TalkSwitch from an electrical outlet to the TalkSwitch unit. Never use a power adapter other than the one that came in the TalkSwitch package.

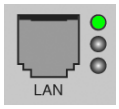
To configure the TalkSwitch unit, connect the unit via ethernet, USB or Internet to the PC with the TalkSwitch management software installed.

The computer connection is only needed to change the configuration settings. It can remain disconnected at other times. If you encounter difficulties opening the TalkSwitch management software, check that all your wires and plugs are securely connected.

Ethernet connection

An Ethernet cable is supplied with specific unit models. This connection requires a LAN and a PC on the same switch. Any PC on the LAN can configure the unit.

The top LED on the back panel beside the LAN port indicates a connection has been established with a switch or hub.



A PC can connect directly to the TalkSwitch LAN port using a 'cross-over' cable equipped with RJ-45 connectors.

You can use a PC that has a wireless connection to the LAN. If the router has uPNP enabled, the TalkSwitch management software can auto-discover and connect to the TalkSwitch unit. If the router does not have uPNP enabled, the TalkSwitch management software cannot auto-discover the TalkSwitch unit. However the software can connect to the TalkSwitch unit if you specify its IP address after starting the software.

Sharing computer data and TalkSwitch voice data, on the same hub can cause voice degradation during high traffic periods. For best results, place TalkSwitch units on a switch, or on their own hub.

1. Connect your PC to the hub or switch using an Ethernet cable.
2. Connect the TalkSwitch unit to your hub or switch using the supplied Ethernet cable.

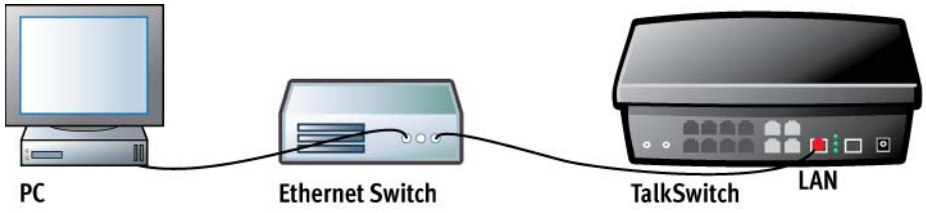


Figure 6: Ethernet connection

USB connection

A USB cable is supplied with specific models. The PC and TalkSwitch unit must be in close proximity.

1. Connect the cable to a USB port on your PC and to the USB port on the back of TalkSwitch.

Internet connection

An Ethernet cable is supplied with specific unit models.

The top LED on the back panel beside the LAN port indicates a connection has been established with a switch or hub.

1. Connect the TalkSwitch unit to your hub or switch using the supplied Ethernet cable.
2. To support remote management over IP, map the TCP/IP 9393 port from your firewall to TalkSwitch. Please refer to the manual for your router/firewall to activate port forwarding.

CONNECTING TELEPHONE LINES

Advisory: In order to minimize disruption to your business, you may want to configure TalkSwitch before connecting it to your telephone lines and local extensions.

The TalkSwitch unit will perform call handling, and will route all incoming calls to the extensions. Call handling includes call forwarding, transferring, voicemail, etc. Telephone lines are not directly connected to any of the local extensions. The TalkSwitch configuration determines how calls are routed to local extensions.

1. Connect the TalkSwitch unit to the telephone company's line sockets using the 2-wire single-line telephone cables provided, beginning with L1. Take note of which telephone line is connected to each line port. This information will be used in the configuration section.

   If you have two telephone lines out of one phone jack, you will require a dual-to-two-single-line adapter or replace the two-line cord with two single-line cords.

Caution: The TalkSwitch line jacks are sensitive to high-voltage spikes from lightning. If you live in an area where electrical storms occur regularly, we recommend you protect TalkSwitch by plugging the telephone cords coming from the TalkSwitch line jacks to a surge protection device connected to the telephone lines.

CONNECTING LOCAL EXTENSIONS

A local extension is an analog extension, IP extension or external IP extension. An analog extension is a device (standard phone, cordless phone, fax machine or modem) connected to extension jack E1–E8. An IP extension is an IP phone connected via LAN. An external IP extension is an IP phone connected via the Internet, and requires the TalkSwitch unit to have VoIP capability.

Connect a single-line device to an extension jack the same way you would plug it into a standard telephone wall jack.

UNITED KINGDOM — To connect telephones with BT style plugs, it is recommended that extensions are wired through a master socket or an in-line adapter with a capacitor. This also ensures compatibility with some telephones that require 3-wire connections for the telephone to ring.

Connecting a single-line analog phone

This is the basic, simple set-up. All functionality including transferring calls, forwarding calls and voicemail, is handled by the unit. Even the most basic of phones, once connected, will inherit these functions.

Each phone is connected to an extension jack. Calls on extension 111 ring the phone connected to jack E1. Calls on extension 112 ring the phone connected to jack E2 and so on.

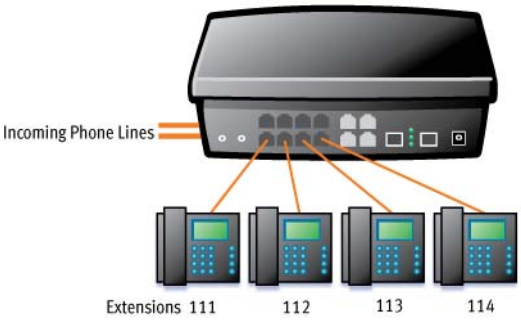


Figure 7: Single-line phones

Connecting a two-line analog phone

You can plug a two-line phone into two extension jacks of the TalkSwitch unit. You can then configure each telephone line to ring a specific extension. For example, line 1 can ring extension 111, which will show Line 1 as active. Line 2 can ring extension 112, which will show Line 2 as active and so on.

Figure 8 shows extensions 111 and 112 connected to a two-line phone that uses separate jacks for each line.

Figure 8: Two-line phone

If your two-line phone does not have separate jacks, you can use a splitter to join the two lines as shown in *Figure 9*.

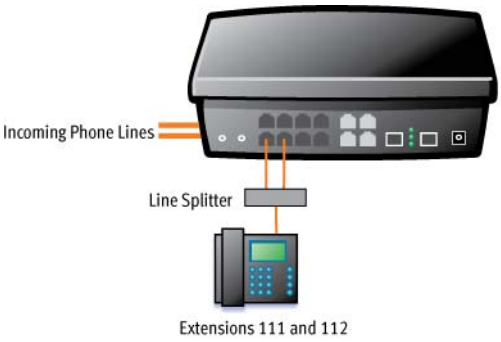


Figure 9: Two-line phone on splitter

A four-line phone can use two splitters to join the four lines as shown in *Figure 10*.

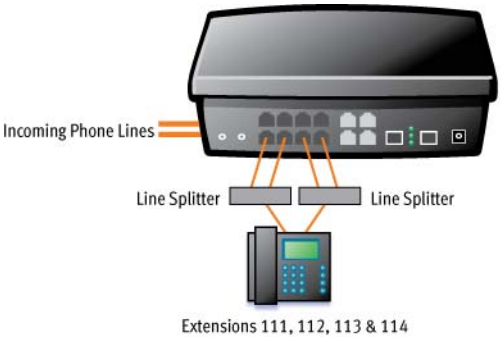


Figure 10: Four-line phone on two splitters

IP Extension

Connecting an IP phone

Plug the IP phone into your switch/hub and configure it as described in *Adding IP phones* on page 81. If you are using an unsupported IP phone, refer to its user guide for configuration instructions.

Connecting a modem

Plug the modem's telephone cable into a unit extension jack. Your modem is now an extension that can access all lines and take advantage of call routing features.

A modem connected to a TalkSwitch unit operates at up to 28 kbps.

If you don't want to change your dial-up settings for the modem, enable Direct Line Access for the extension associated to the modem. For more information, see *Direct line access* on page 61.

Connecting a fax machine

If you are connecting a fax machine to a TalkSwitch unit, plug it into an extension jack.

TalkSwitch provides you with three options to configure your fax machine. Choose the best option for your office.

Option 1 — Dedicated fax line

You may already have an incoming dedicated fax line. You can leave this line directly connected to your fax machine. Connect the remaining telephone lines to your unit. Telephone lines will be handled by the unit and your fax will work the way it has always worked.

After installing TalkSwitch, the fax machine remains on its own dedicated line.
The remaining telephone lines are shared by the phones.

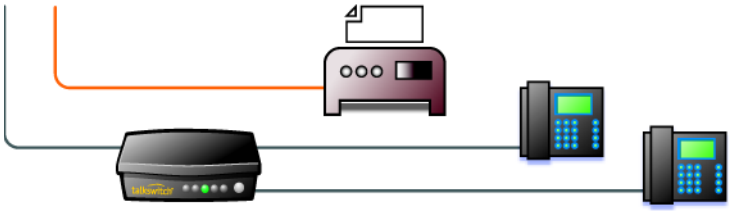


Figure 11: Dedicated fax line

Advantages: It's easy to set up and you don't need to change the way your fax handles calls.

Disadvantages: The dedicated fax line cannot be shared with the other phone devices (i.e. you cannot make an outbound voice call on your fax line). The dedicated fax line costs money, and may not be used as frequently as it could be.

Option 2 — Distinctive ring

Distinctive ring is a service provided by your phone company. A second phone number is added to a telephone line. A call to either phone number will ring the same line, but with different ring patterns. You can use one ring pattern to indicate a fax.

The TalkSwitch unit detects the ring pattern. Calls to the main number are automatically routed to the phones. Calls to the distinctive ring number are automatically routed to the fax machine.

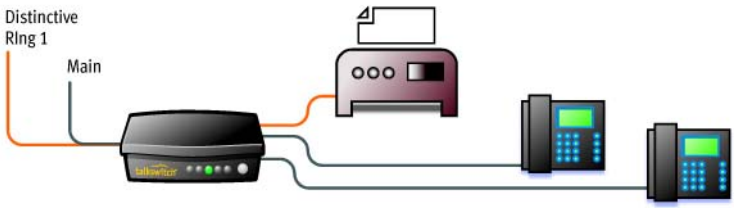


Figure 12: Distinctive ring

Advantages: This option does not require a second physical line for faxes. More telephone lines can be added as you grow. You can keep the same fax arrangement.

Disadvantages: Your phone company may charge a fee for the distinctive ring. If you choose this option, you will require the distinctive ring service from your telephone company.

Option 3 — Automatic fax detection via the auto attendant

The auto attendant can detect incoming fax calls and route them accordingly. To enable this, an auto attendant must answer all incoming calls. It will listen for a fax tone (CNG) to determine if the call is an incoming fax. If so, the call is routed to the fax machine. You must create an auto attendant to use this method.

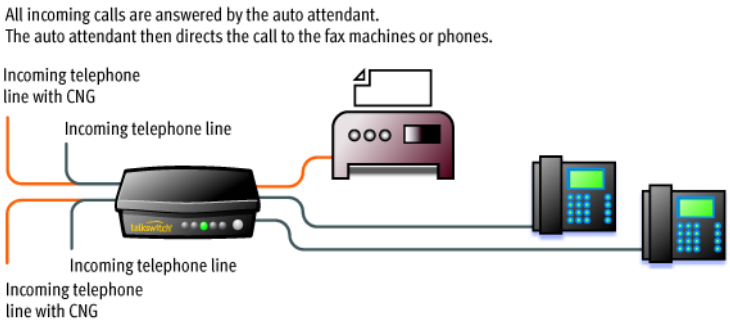

All incoming calls are answered by the auto attendant.
The auto attendant then directs the call to the fax machines or phones.
Incoming telephone line with CNG
Incoming telephone line
Incoming telephone line
Incoming telephone line with CNG

Figure 13: Automatic fax detection via auto attendant

Advantages: Calls are automatically handled for you. Distinctive ring service is not required. Any telephone line can receive a fax or a call, making better use of your resources.

Disadvantages: Some older fax machines do not emit a fax tone (CNG). Therefore, the auto attendant cannot detect incoming faxes from those devices. If your auto attendant volume is set too high or if your telephone lines are noisy, the auto attendant may have difficulty detecting the incoming fax tone and therefore not route the call to the fax machine. You can solve this problem by adjusting the volume. For more information, see *Telephone Lines* > *Audio* on page 194.

CONNECTING TO THE PA (PUBLIC ADDRESS) JACK

The PA jack can be connected to a PA system for overhead paging, screening voicemail and playing music on hold. The PA jack requires a 1/8" (3.5 mm) mono phono connector. If your system has multiple TalkSwitch units, use a splitter to connect each TalkSwitch unit to the PA system.

SETTING UP MUSIC ON HOLD

Callers tend to react more positively to waits or transfers if they hear music. You can also play an advertisement or provide information about your company.

External audio source

The MUSIC jack is designed to support an external audio source like a CD player, tape player or PC media player.

We recommend that any external audio source be connected to the TalkSwitch unit using an approved Line Isolation Unit (e.g. A-Tick ( ) approved in AU).

Advantages: The length of the recording is not limited by the amount of memory in the TalkSwitch units. All of the TalkSwitch memory is available for voicemail and auto attendants. Different TalkSwitch units can have different music on hold.

Disadvantages: Extra dedicated equipment must be connected and running constantly. A separate sound source is required for each unit.

1. Connect the audio source via its headphone output to the MUSIC jack. Use a cable with a 1/8" (3.5 mm) mono phono connector.

2. If your system has multiple TalkSwitch units, either use a different audio source for each unit, or use a splitter to connect the one audio source to each unit.

Internal audio file

You can load an audio .wav file into the TalkSwitch unit. If you have more than one unit connected by LAN, the .wav file is automatically copied to each unit.

Advantages: No additional equipment is required. All units use the same file, which provides seamless music for calls transferred across the system.

Disadvantage: The length of the recording is limited by the amount of memory in the TalkSwitch units. Less time is available for voicemail and auto attendants.

The recording time for internal music on hold, voicemail, and the auto attendants is shared on the TalkSwitch unit. You can expand the built-in memory using memory cards.

NETWORKING TALKSWITCH UNITS ON A LAN

Up to four units can be networked together over a LAN to increase the number of telephone lines and extensions in your phone system.

Connecting TalkSwitch units to a LAN

You can plug your unit(s) into your existing LAN or operate it on its own LAN with an Ethernet switch. Ensure the firmware on all units is updated before adding them to the LAN.

Check the support section at **http://global.talkswitch.com** for the latest updates.

TalkSwitch unit enclosures are not designed for stacking. We recommend wall-mounting units to maximize airflow and keep the units from overheating.

Ethernet switch

An Ethernet switch provides direct communication between units, thus keeping the voice-over-LAN data isolated from other data on the network.

1. Connect up to four units to an Ethernet switch.
2. Connect a computer to the Ethernet switch.
3. Power up and configure the units.
4. After configuring the units, connect the IP extensions to the Ethernet switch.

It is important that you place the IP extensions on the same subnet as TalkSwitch. For example, if your TalkSwitch unit has the IP address 192.168.1.200, your IP extensions should use IP addresses in the range 192.168.1.xxx that do not conflict with other IP addresses.

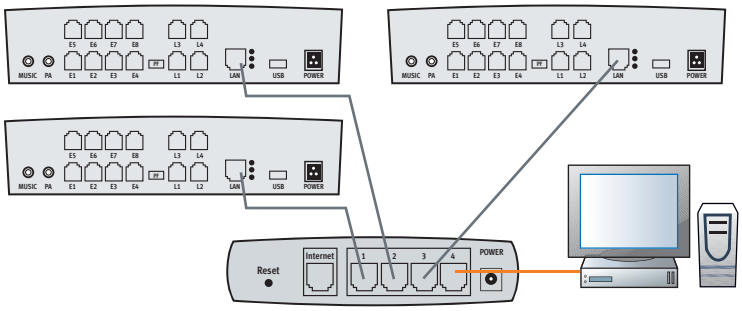


Figure 14: Three units connected via an Ethernet switch

Power up all the TalkSwitch units

Connect the provided AC adapters to each unit and plug the adapters into available power outlets. Turn the units on by pressing the Power button on the front of each unit. The lights on the front panels of the units will flash. This means they are powered up and connected to the LAN, but their ID numbers are not set.

Setting or changing the unit ID

Units are shipped from the manufacturer with unit ID 1. If two or more units are placed on the same LAN with the same unit ID number, the newest unit will adopt the next available higher unit ID. For example, when a new unit is added to a system with unit IDs 1 and 2, the new unit automatically assumes unit ID 3.

You can specify the unit ID:

1. Connect an analog phone to one of the extension jacks on the unit.
2. Press # to enter command mode.
3. Dial the system password + #.
4. Now, you have the following options:

Keys	Result
00 + #	Responds with the unit ID number of that particular unit.
01 + #	Sets the unit ID to 1.
02 + #	Sets the unit ID to 2.
03 + #	Sets the unit ID to 3.
04 + #	Sets the unit ID to 4.

The system will confirm the new ID is accepted. When the front panel Data light is on solid, the units are ready for network use.

How unit IDs affect system extension numbers

After successfully changing each unit ID to a unique number, it can be identified by the network. More importantly, each local extension, remote extension, and voice mailbox has a unique number. Note that dialing 101–108 will ring local extensions 111–118.

	Unit ID 1	Unit ID 2	Unit ID 3	Unit ID 4
Local extensions (analog/IP)	111–118	121–128	131–138	141–148
Local extensions (IP only)	151–158	161–168	171–178	181–188
Remote extensions	210–219	220–229	230–239	240–249
Local mailboxes	111–118 151–158	121–128 161–168	131–138 171–178	141–148 181–188
Remote mailboxes	210–219	220–229	230–239	240–249
General mailboxes	410–419	420–429	430–439	440–449

Keep track of the lines and extensions

To keep track of the extensions and telephone lines connected to each unit, label each unit with its unit ID. It simplifies matters when you need to add or remove extensions and lines.

Optimizing the system for networked use

TalkSwitch units have been designed to operate optimally when in a networked state. Below are a few items that have been designed for better network use.

Configuration settings

All units are cloned with identical settings. If a unit has disappeared off the network (adapter unplugged, LAN disconnected, LAN failure, etc.), the system can still handle the calls since it retains the configuration settings of the missing unit. If an extension or a voice mailbox cannot be reached, the caller hears a system prompt: *"The extension you are trying to reach is currently unavailable, please try again later."*

Outgoing line hunt groups

When two or more units are on a LAN, the system tries placing calls out on the same unit the call originates from. This avoids using a line across the LAN on another unit, which helps to minimize the LAN traffic and optimizes the opportunity for all inbound calls to connect across the LAN if required.

Auto attendants

There are a total of nine auto attendants that are shared by all units on the LAN. When an auto attendant is recorded on any unit, it is automatically copied to all other units on the LAN. This design minimizes LAN traffic and provides functionality back-up in case a unit or units have disappeared off the network (adapter unplugged, LAN disconnected, LAN failure, etc.). In this event, all units can still answer inbound calls with the same auto attendant messages.

Voicemail

All local extension and remote extension voicemail data is stored on the unit the extension is connected to. For example, all greetings, dial-by-name directory names and voicemail messages for extensions 121–128, 161–168, 220–229 and mailboxes 420–429 are stored on unit 2. If a unit is completely filled with voicemail messages, the system will not accept any more messages for its extensions. However the other units will accept messages for their extensions. If you need more memory, TalkSwitch memory cards can be purchased from your reseller. To view the memory usage of each TalkSwitch unit and each mailbox, choose **Tools > Memory Usage** or choose **Tools > Voicemail Manager > Mailbox Status**.

Special considerations when connecting multiple units to a LAN

Multiple TalkSwitch units on a LAN will function as a single phone system. Some features are expanded as you add units, and other features are not.

As you add more units, the number of telephone lines, local extensions, VoIP lines, remote extensions and voice mailboxes will increase.

For example, if you have a 480vs and a 244vs connected on the same LAN, the system can support 6 telephone lines, 32 local extensions, 4 VoIP lines, 20 remote extensions, 32 local voice mailboxes, 20 remote voice mailboxes, and 20 general voice mailboxes.

The features listed below are not affected by the number of units on the LAN, so capacity will not increase as you add units.

Feature	Description
Auto Attendant	There are nine auto attendants.
Ring Groups	There are ten ring groups (300–309).
Park Orbits	There are ten park orbits (500–509).
Music on Hold	You can add music on hold to your system by loading an internal music file. The music file is copied to each TalkSwitch unit, so the music on hold will be the same for each line of each unit.
Public Address (PA)	Each unit has a PA output for connection to a public address system. The PA output for a TalkSwitch unit is only accessible from extensions connected to that unit. To connect all outputs to a single amplifier, you will require an adapter.

INCREASING MEMORY CAPACITY

The recording time for internal music on hold, voicemail, and the auto attendants is shared on the TalkSwitch unit. You can expand the built-in memory using memory cards. Contact your reseller to purchase memory upgrades.

TalkSwitch memory cards

TalkSwitch memory cards are specifically formatted for use with TalkSwitch units. They allow you to add additional voicemail message capacity to your unit. Voicemail memory is used to store greetings and messages left in voice mailboxes. It is also used to store music on hold, auto attendant greetings and announcements.

Figure 15: Memory card

Installing a memory card

1. Insert the memory card into the memory slot with the gold contact face down and the “This side up” label showing. You don’t have to turn the unit off or disconnect any of the telephones.

2. Ensure the *Total Unit Time* has increased in the **Mailbox Data** window. See *Memory Usage* on page 178.

Figure 16: Memory card slot

UPGRADING TALKSWITCH UNITS

As your business grows, your TalkSwitch unit(s) may need upgrading. You have the option of upgrading your units to VoIP, as well as adding telephone lines and extension jacks. The upgrade kit consists of a top board you add to your existing TalkSwitch unit, as well as a guide containing detailed, step-by-step instructions for the upgrade procedure.

Purchase VoIP modules and upgrade boards at your reseller or visit the TalkSwitch website for more details.

Note that in some cases it may be easier and more cost effective to buy another TalkSwitch unit than to upgrade an existing unit.

Upgrade	TalkSwitch unit	Board	TalkSwitch equivalent
Add four VoIP lines	240vs 280vs 480vs 840vs 48-CA	4plus	244vs 284vs 484vs 844vs 48-CVA
Add eight VoIP lines	240vs 280vs 480vs 840vs 48-CA	8plus	248vs 288vs 488vs 848vs 48-CVAv2
Add six telephone lines	240vs 244vs 248vs	600plus	840vs 844vs 848vs
Add four extension jacks	240vs 244vs 284vs	40plus	280vs 284vs 288vs
Add two telephone lines and four extension jacks	240vs 244vs 248vs 24-CA	240plus 24-CA to 48-CA top board	480vs 484vs 488vs 48-CA

ENABLING LICENSED ADD-ONS

It is possible to enable the following features on your TalkSwitch:

- Attendant Console software
- Call Reporting software
- Extended Warranties (available in Canada and US only)

For instructions on enabling TalkSwitch licensed add-ons, see *Licensed Add-Ons* on page 56.

VERIFYING THE CONNECTIONS

If you are having problems, they could be due to loose cables. Check the following:

1. Are your telephone cables connected to a wall jack at one end and the unit at the other? Ensure the devices you are using as extensions (telephones, faxes, etc.) are plugged into the extension jacks located on the back panel of the unit. Do not plug your extensions directly into a wall jack unless that jack is wired to a unit extension jack.
2. Check the USB or LAN connection between TalkSwitch and your PC.
3. Ensure the unit AC adapter is plugged into a working power outlet and the unit is turned on. Make sure you are using the AC adapter supplied with your unit. This adapter cannot be substituted with any other adapter. If your adapter appears to be defective, please contact your TalkSwitch dealer to order the proper replacement.

If networked, ensure you have the latest firmware loaded on all units.

CONNECTING A TALKSWITCH UNIT TO A PBX

Connecting a TalkSwitch unit to a PBX allows the TalkSwitch unit to answer calls with an auto attendant. The caller can dial **7** followed by an extension number to reach an extension on the PBX.

1. Connect one or more PBX extensions to the TalkSwitch telephone line jacks (one PBX extension per line).
2. Configure the PBX to route incoming calls to the PBX extensions that are connected to the telephone line jacks of the TalkSwitch unit.
3. Configure the auto attendant to answer after a specified number of rings.
4. Record the auto attendant messages. They should list the extensions with the digit 7 in front of them.
5. Set the *PBX extension length*. See *From auto attendant tab* on page 186.

For example, if you have chosen the PBX extension length to be 3 digits and John Smith's extension is 111 on the PBX, the auto attendant should say: *"To speak with John Smith, dial 7111."* When the caller dials 7111, the TalkSwitch unit puts the caller on hold and transfers them to extension 111 on the PBX.

This feature only works with PBXs that use hook flash as a transfer method. Please contact your PBX manufacturer for transfer details.

CHAPTER 2

TALKSWITCH CONFIGURATION

INTRODUCTION

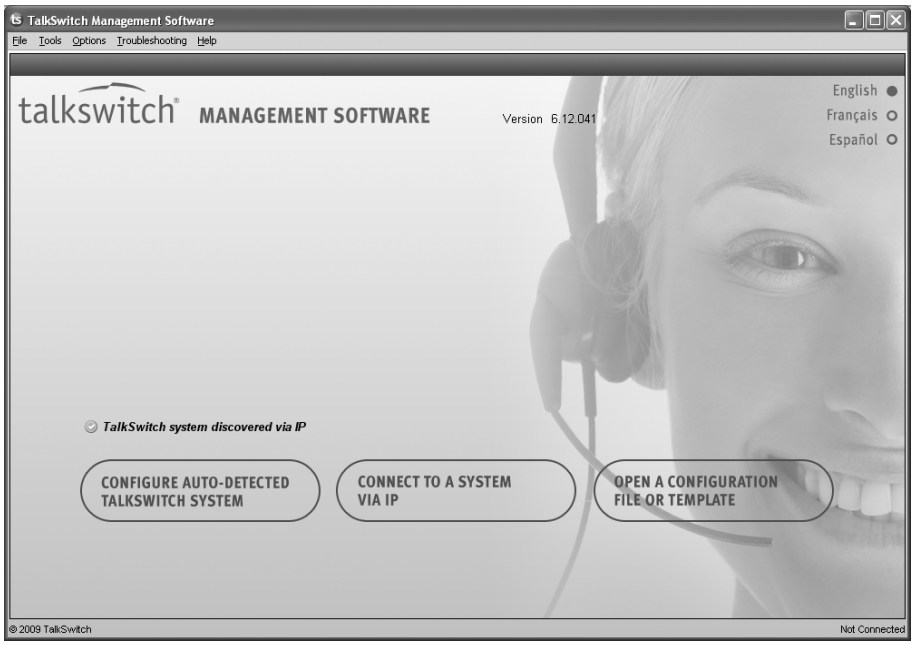
This chapter contains detailed information about all the features in the TalkSwitch management software, with step-by-step instructions on how to customize these features to best suit your needs.

STARTING THE TALKSWITCH MANAGEMENT SOFTWARE

When a unit is being configured, it is locked to prevent other computers or persons using a phone from configuring changes at the same time. If you leave the software open for longer than one hour, the unit unlocks itself to allow configuration changes.

1. Press the Power button to turn on your TalkSwitch unit.
2. Turn on your computer.

3. Double-click the **TalkSwitch** icon on your desktop to open the TalkSwitch management software. You can also click **Start** > **Programs** > **TalkSwitch 6.12** > **TalkSwitch Configuration 6.12**. The **Configuration Selection** page appears, and the software attempts to detect your TalkSwitch unit.



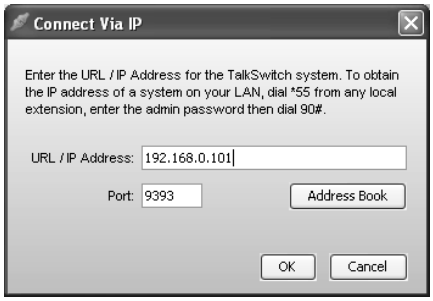
4. Select your language. You can select *English*, *Français* or *Español*.

5. Once the software detects your TalkSwitch unit, click **Configure Auto-Detected TalkSwitch System**. The software loads the configuration from the TalkSwitch unit, and the **About** page appears.

6. If the software was unable to detect your TalkSwitch unit, check that all your wires and plugs are securely connected, and then click **Retry Auto-Discovery**. The software loads the configuration from the TalkSwitch unit, and the **About** page appears.

7. If auto-detection does not work:

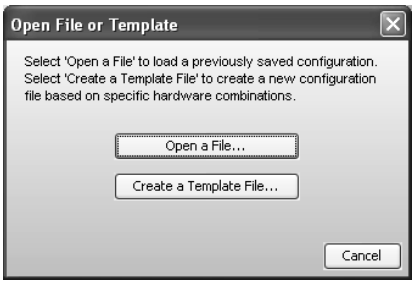
   a) Click **Connect to Different System**. The **Connect Via IP** window appears.


Connect Via IP
Enter the URL / IP Address for the TalkSwitch system. To obtain the IP address of a system on your LAN, dial *55 from any local extension, enter the admin password then dial 90#.
URL / IP Address: 192.168.0.101
Port: 9393
Address Book
OK
Cancel

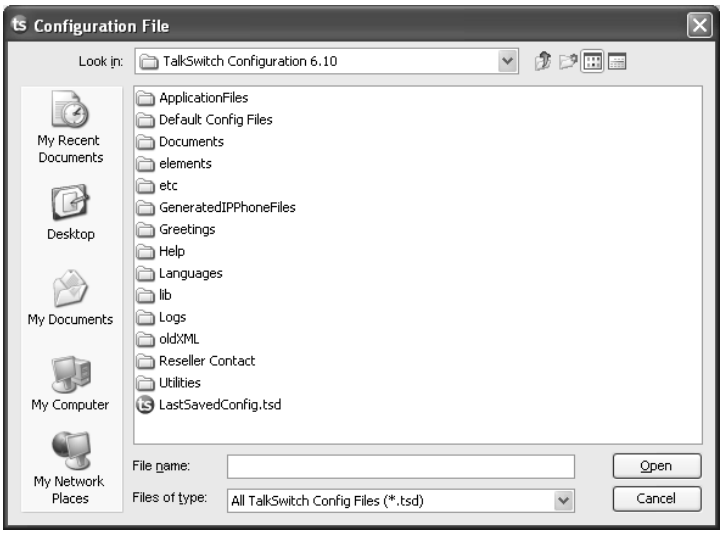
   b) Enter the *URL / IP address* and *Port* of the TalkSwitch unit, and then click **OK**. This is the public IP address or FQDN of the TalkSwitch system. The software loads the configuration from the TalkSwitch unit, and the **About** page appears.

8. To open a configuration file:

a) Click **Open a Configuration File or Template**. The **Open File or Template** window appears.


Open File or Template
Select 'Open a File' to load a previously saved configuration.
Select 'Create a Template File' to create a new configuration file based on specific hardware combinations.
Open a File...
Create a Template File...
Cancel

b) Click **Open a File**. The **Configuration File** window appears.

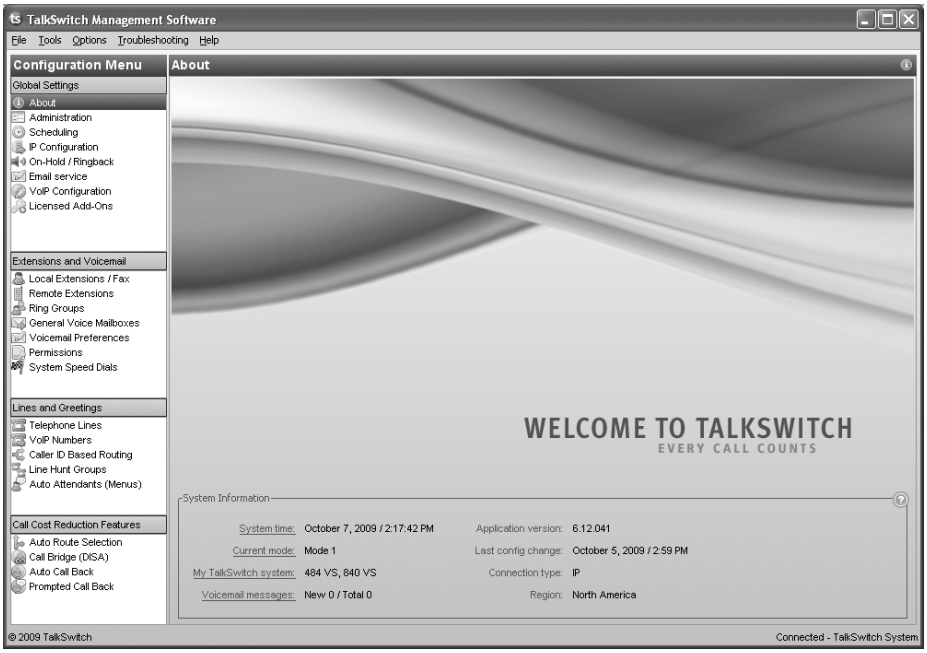


c) Browse to the .tsd file, and then click **Open**. The software opens the configuration file.

ABOUT

The **About** page displays and allows you to change system information.

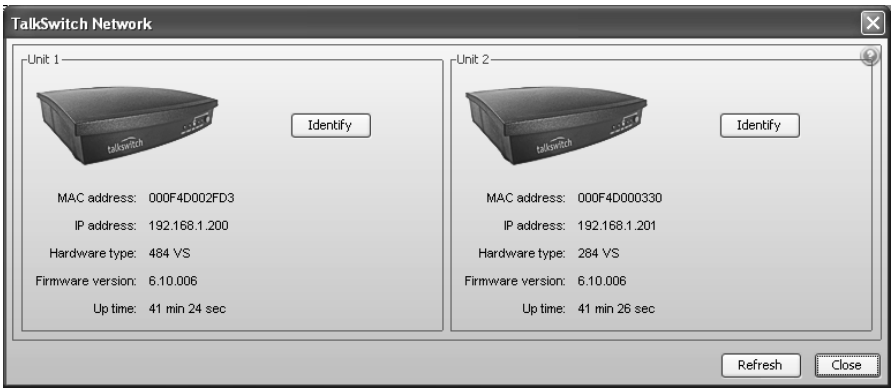
1. Select the **About** page.



The *System Information* area shows:

- The current time and date programmed into the TalkSwitch system. Clicking the **System time** link displays the **TalkSwitch Date and Time Properties** window. It allows you to change the date, time, time zone, and NTP server programmed into the TalkSwitch system. See *System time* on page 31.
- The current mode. A mode is a period of time when the TalkSwitch system uses a particular call handling setup for incoming calls. Mode 1 is typically office hours, and Mode 2 is typically evening and weekend hours. Holiday Mode is when your office is closed for a statutory holiday or shutdown. Clicking the **Current mode** link displays the **Change Mode** window. It allows you to change the mode of the TalkSwitch system. See *Change mode* on page 30.

- The model numbers of the TalkSwitch units in the TalkSwitch system. Clicking the **My TalkSwitch system** link displays the **Talkswitch Network** window. It shows the MAC address, IP address, model number, firmware version, and length of operation for each TalkSwitch unit.



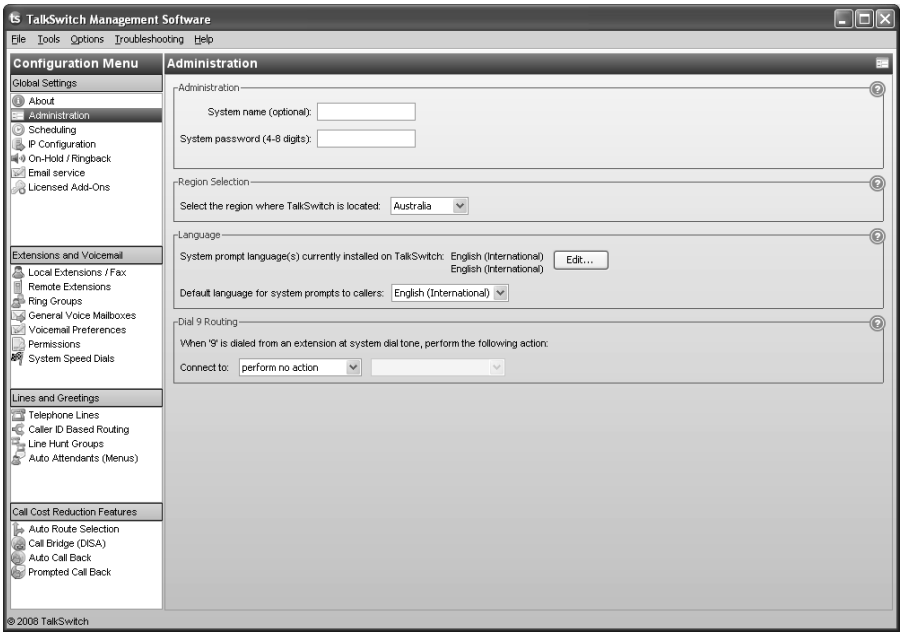
  To identify a TalkSwitch unit, click the **Identify** button. The **Identify** button will change to a **Stop** button. All the lights on the front panel will start flashing. Click the **Stop** button to end the flashing.

- The number of new voicemail messages in the TalkSwitch system, and the total number of voicemail messages. Clicking the **Voicemail messages** link displays the **Voicemail Memory Usage** window. It shows message statistics for each voice mailbox. See *View mailbox data* on page 125.
- The version number of the TalkSwitch management software.
- When the configuration was last saved to the TalkSwitch system.
- The type of connection to the TalkSwitch system.
- The region of the TalkSwitch system.

ADMINISTRATION

The **Administration** page allows you to set up the system name, system password, region,language, and how to route the call if a user dials an operator number.

1. Select the **Administration** page.



Note: The *Region Selection* area is only displayed in certain regions. Depending on the region, an operator may be dialed using 9 or 0.

Administration

The *Administration* area allows you to set up the system name and system password.

CAUTION: Use a system password, otherwise the system will be vulnerable to configuration changes, misuse and/or lock-out by callers or users. If the router has port 9393 mapped to the TalkSwitch unit for remote configuration, the system will also be vulnerable to anyone on the Internet.

Change the system password frequently to prevent unauthorized users from making calls or changing the configuration.

1. Optionally enter the *System name*. This should be the company name, or a shortened form suitable for use as caller ID during VoIP calls. If you enter a system name, you will require it when logging in through a browser for status or call detail record (CDR) information. If you omit the system name, you can enter **talkswitch** when logging in through a browser.

2. Enter the *System password*. It has to be a 4- to 8-digit numeric password, so you can also enter it on a touchtone phone.

   The system password allows access to the configuration. It is entered when you start the TalkSwitch management software, and when you access the configuration through a local extension or outside phone.

Region selection

The *Region Selection* area (displayed only in certain regions) allows you to select the country where your TalkSwitch unit will operate.

Language

The *Language* area displays the language loaded into the TalkSwitch system, and allows you to load and remove language files.

1. To change the language files loaded into the TalkSwitch system, click **Edit**. The **Language File Management** window appears, listing loaded language files.

   a) To load a language file, click **Add**, and then select the language file.

   b) To remove a language file, select the language, and then click **Remove**.

2. Select the default language for prompts heard by callers and users in the *Default language for system prompts to callers* list.

Dial 9 or 0 routing

The *Dial 9 routing* area (or *Dial 0 routing* area in some regions) allows you to select how the system will route the call if a user dials **9** or **0**. For example, you can configure the system to connect the user to the receptionist at extension 114.

1. Select the action in the *Connect to* list. Choices are:

   - *perform no action* — Does not connect to a resource.
   - *go to voicemail* — Connects to the selected voicemail.
   - *go to local extension* — Connects to the selected local extension.
   - *go to remote extension* — Connects to the selected remote extension.
   - *go to ring group* — Connects to the selected ring group.
   - *play announcement* — Plays the selected announcement.
   - *go to auto attendant* — Connects to the selected auto attendant.
   - *queue at ring group* — Connects to the call queue of the selected ring group.
   - *lookup name directory* — Accesses the dial-by-name directory.
   - *go to VoIP location* — Connects to the selected VoIP number within the TalkSwitch VoIP network.

2. Select the resource. Depending on the action, resources are voice mailboxes, extensions, announcements, auto attendants, or VoIP numbers within the TalkSwitch VoIP network.

SCHEDULING

A mode is a period of time when the TalkSwitch system uses a particular call handling setup for incoming calls. Mode 1 is typically office hours, and Mode 2 is typically evening and weekend hours. Holiday Mode is when your office is closed for a statutory holiday or shutdown.

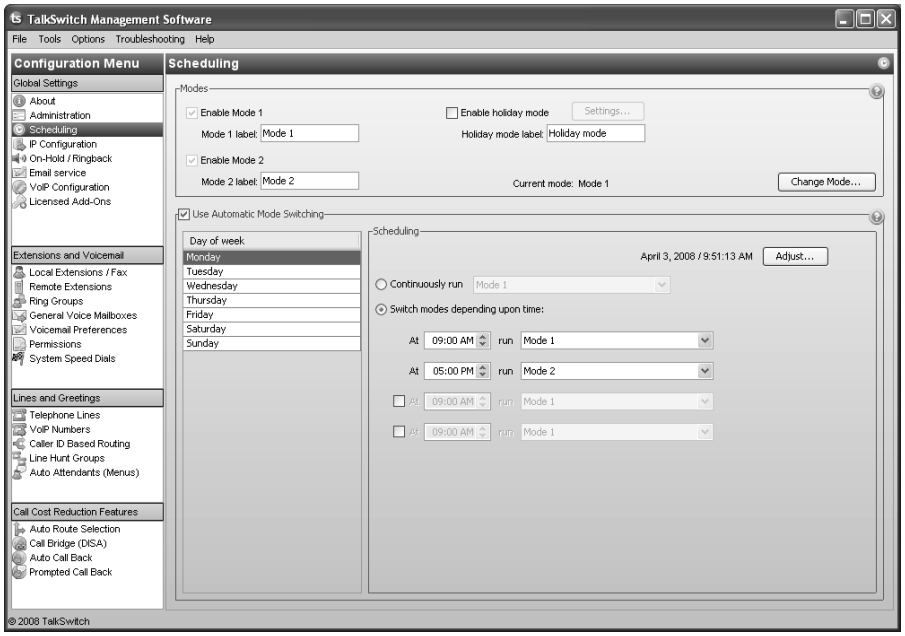
The TalkSwitch system can use a different call handling setup for each mode, and can automatically change mode with the time of day, day of week and on holidays. This way it can handle calls in different ways according to a schedule.

The **Scheduling** page allows you to set up Mode 1, Mode 2 and Holiday Mode. You can also adjust the time and date, and can switch modes manually.

Ensure the TalkSwitch management software is closed during automatic mode switching, or if an administrator is manually switching modes using a phone. In these cases the mode cannot switch while the TalkSwitch management software is open.

Note that the mode can switch while the TalkSwitch management software is open, if the user is manually switching modes using the software, as described later.

1. Select the **Scheduling** page.



Modes

The *Modes* area allows you to enable, name and switch modes, and displays the current mode.

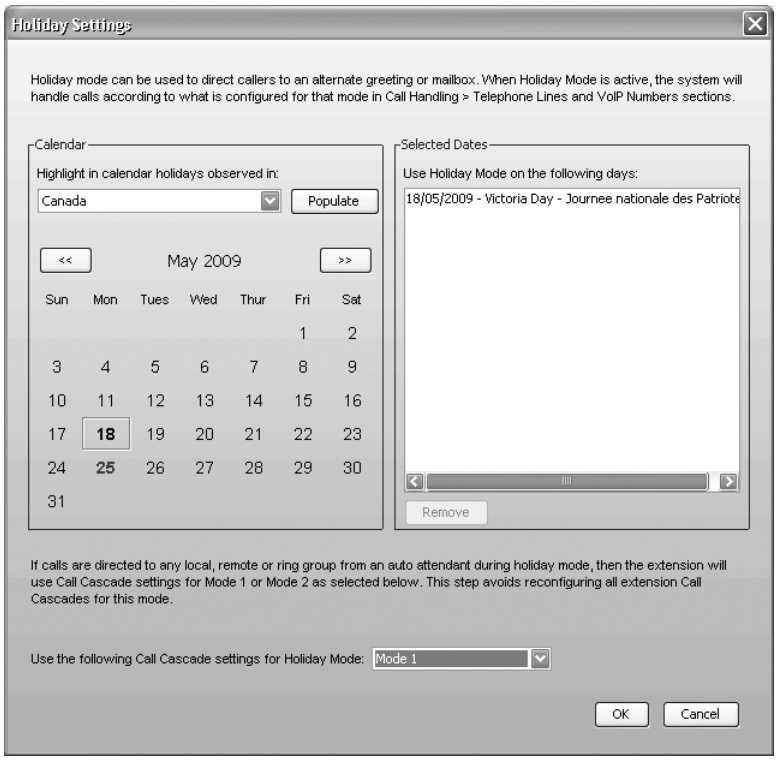
1. Enter the *Mode 1 label* and the *Mode 2 label*. Note that Mode 1 and Mode 2 are always available, so you cannot disable the *Enable Mode 1* and *Enable Mode 2* checkboxes.

2. Set up Holiday Mode to handle incoming calls differently during a holiday:

   a) Select the *Enable holiday mode* checkbox to enable Holiday Mode. The window enables the **Settings** button.

   b) Enter the *Holiday Mode label*.

c) Click **Settings**. The **Holiday Settings** window appears. The calendar shows the current date in green.


Holiday Settings
Holiday mode can be used to direct callers to an alternate greeting or mailbox. When Holiday Mode is active, the system will handle calls according to what is configured for that mode in Call Handling > Telephone Lines and VoIP Numbers sections.
Calendar
Highlight in calendar holidays observed in:
Canada
Populate
<<
May 2009
>>
Sun Mon Tues Wed Thur Fri Sat
1 2
3 4 5 6 7 8 9
10 11 12 13 14 15 16
17 18 19 20 21 22 23
24 25 26 27 28 29 30
31
Selected Dates
Use Holiday Mode on the following days:
18/05/2009 - Victoria Day - Journee nationale des Patriote
Remove
If calls are directed to any local, remote or ring group from an auto attendant during holiday mode, then the extension will use Call Cascade settings for Mode 1 or Mode 2 as selected below. This step avoids reconfiguring all extension Call Cascades for this mode.
Use the following Call Cascade settings for Holiday Mode:
Mode 1
OK
Cancel

d) Select your country in the *Populate calendar with holidays observed in* list, and then click **Populate**. The calendar will show your country's statutory holidays in grey.

e) Using the calendar, select the month and then click the date of the holiday. The date is added to the *Selected Dates* area, and the calendar will show the date in blue. Click the date again to remove it.

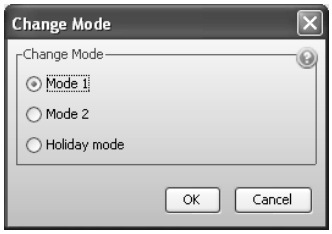
f) Repeat Step e until all required holidays have been added.

g) Select the mode in the *Use the following call cascade settings for Holiday Mode*. Choices are *Mode 1* and *Mode 2*. This setting determines whether extensions will use their Mode 1 or Mode 2 call cascades during Holiday Mode.

Change mode

The *Modes* area displays the current mode.

1. To change mode, click **Change Mode**. The **Change Mode** window appears.



2. Select the mode. The new mode immediately takes effect. You do not have to save the configuration.

You can also change the mode by phone.

1. Ensure the TalkSwitch management software is closed.
2. Pick up a local extension, or remotely call the TalkSwitch system.
3. Enter command mode by either pressing **#** on an analog extension phone, or ***55#** on a TalkSwitch IP phone (note: other brands may use ***55 Send** or ***55 Dial**).
4. Enter the system password, followed by **#**.
5. Enter one of the following commands:
   - **30** + **#** to hear the current mode.
   - **31** + **#** to enable Mode 1.
   - **32** + **#** to enable Mode 2.
   - **33** + **#** to enable Holiday Mode.
6. Press ***** to exit command mode.

Scheduling

The *Scheduling* area allows you to specify when the TalkSwitch system will change modes.

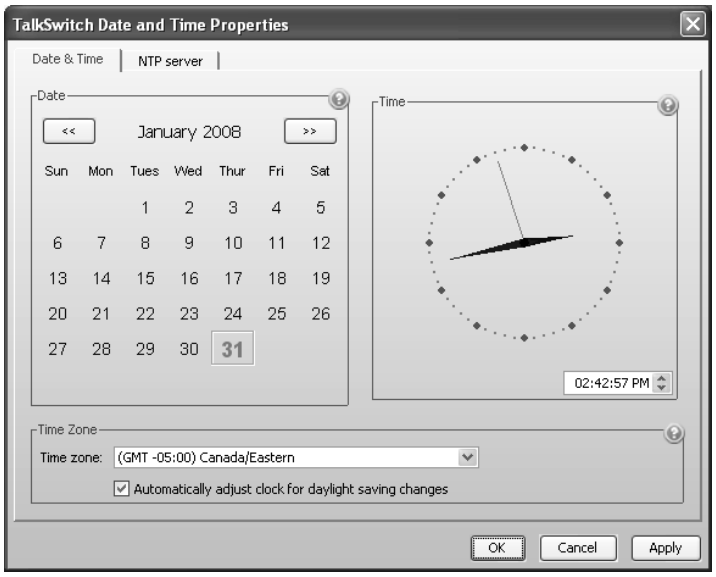
1. Select the *Use automatic mode switching* checkbox to enable the modes. The window enables the *Scheduling* area.
2. Select the day.
3. Select whether to use one mode or multiple modes during that day:
   - Select *Continuously run* to use the selected mode the whole day.
   - Select *Switch modes depending on time* to switch between Mode 1, Mode 2 and Holiday Mode during the day. The window enables the schedule controls.
4. If you selected *Continuously run*, select *Mode 1*, *Mode 2* or *Holiday Mode*.
5. If you selected *Switch modes depending on time*, set when the mode should change. The default settings are:
   - Mode 1 at 9:00 AM on Monday to Friday.
   - Mode 2 at 5:00 PM on Monday to Friday.
   - Mode 2 continuously on Saturday and Sunday.

System time

The *Scheduling* area shows the date and time programmed into the TalkSwitch system. Clicking the **Adjust** button displays the **TalkSwitch Date and Time Properties** window. It allows you to change the date, time, time zone, and NTP server programmed into the TalkSwitch system.

NOTE: You can configure the TalkSwitch system to get the time and date (but not the year or time zone) from caller ID. See *Clock Preferences* on page 189.

1. Click **Adjust**. You can also choose **Options > Set Date & Time**. The **TalkSwitch Date and Time Properties** window appears.



Date & time

The *Date & Time* area allows you to set the date and time programmed into the TalkSwitch system.

1. Select the month and date for the TalkSwitch system.
2. Set the time for the TalkSwitch system.

Time zone

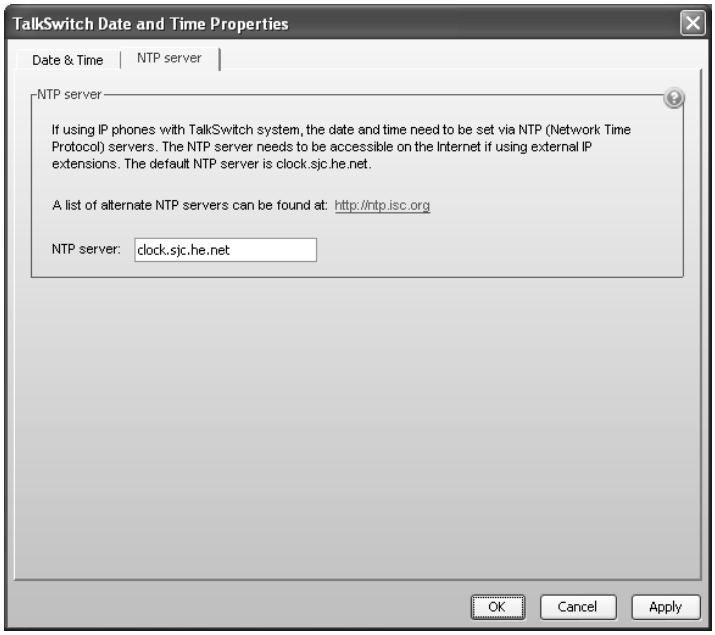
The *Time Zone* area allows you to set the time zone for the TalkSwitch system, and whether daylight savings time is used in your region.

1. Select the time zone for the TalkSwitch system.
2. If your region uses daylight savings time, select the *Automatically adjust clock for daylight saving changes* checkbox.

NTP server

The *NTP Server* area allows you to set the NTP (Network Time Protocol) server for the TalkSwitch system. It provides the same time to the TalkSwitch units and to the external IP extensions.

1. Select the *NTP Server* tab.

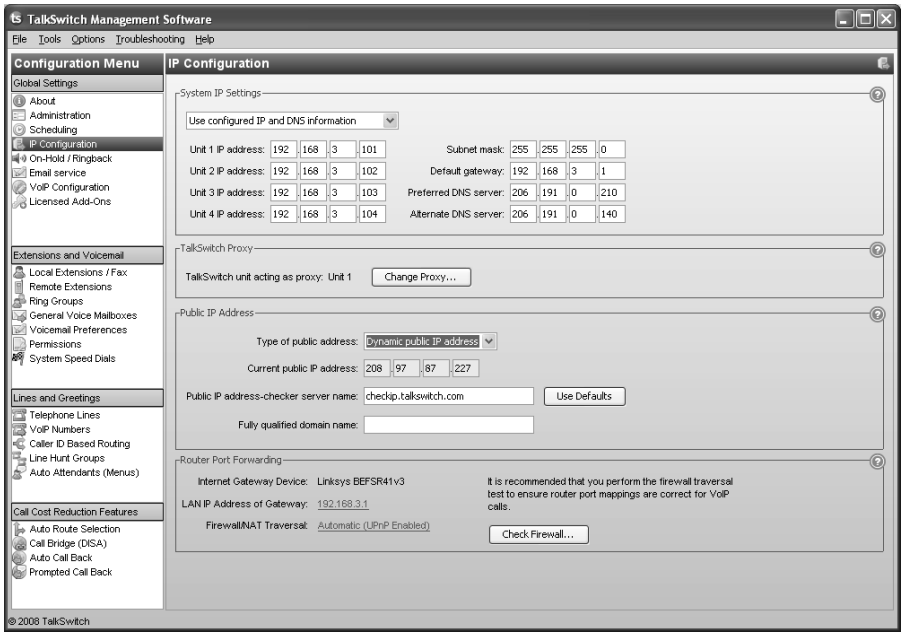

TalkSwitch Date and Time Properties
Date & Time
NTP server
NTP server
If using IP phones with TalkSwitch system, the date and time need to be set via NTP (Network Time Protocol) servers. The NTP server needs to be accessible on the Internet if using external IP extensions. The default NTP server is clock.sjc.he.net.
A list of alternate NTP servers can be found at: http://ntp.isc.org
NTP server: clock.sjc.he.net
OK
Cancel
Apply

2. Enter the *NTP server*.

IP CONFIGURATION

The **IP Configuration** page allows you to set up the TalkSwitch system for Internet communications. The Internet can be used for external IP extensions, a TalkSwitch VoIP network, a subscription to a VoIP service, and remote configuration.

1. Select the **IP Configuration** page.


TalkSwitch Management Software
File Tools Options Troubleshooting Help
Configuration Menu
Global Settings
About
Administration
Scheduling
IP Configuration
On-Hold / Ringback
Email service
VoIP Configuration
Licensed Add-Ons
Extensions and Voicemail
Local Extensions / Fax
Remote Extensions
Ring Groups
General Voice Mailboxes
Voicemail Preferences
Permissions
System Speed Dials
Lines and Greetings
Telephone Lines
VoIP Numbers
Caller ID Based Routing
Line Hunt Groups
Auto Attendants (Menus)
Call Cost Reduction Features
Auto Route Selection
Call Bridge (DISA)
Auto Call Back
Prompted Call Back
IP Configuration
System IP Settings
Use configured IP and DNS information
Unit 1 IP address: 192 168 3 101
Unit 2 IP address: 192 168 3 102
Unit 3 IP address: 192 168 3 103
Unit 4 IP address: 192 168 3 104
Subnet mask: 255 255 255 0
Default gateway: 192 168 3 1
Preferred DNS server: 206 191 0 210
Alternate DNS server: 206 191 0 140
TalkSwitch Proxy
TalkSwitch unit acting as proxy: Unit 1
Change Proxy...
Public IP Address
Type of public address: Dynamic public IP address
Current public IP address: 208 97 87 227
Public IP address-checker server name: checkip.talkswitch.com
Use Defaults
Fully qualified domain name:
Router Port Forwarding
Internet Gateway Device: Linksys BEFSR41v3
LAN IP Address of Gateway: 192.168.3.1
Firewall/NAT Traversal: Automatic (UPnP Enabled)
It is recommended that you perform the firewall traversal test to ensure router port mappings are correct for VoIP calls.
Check Firewall...
© 2008 TalkSwitch

System IP settings

The *System IP Settings* area shows IP addresses. By default, *Obtain IP and DNS information automatically* is selected and the area shows IP addresses from the router.

1. Change *Obtain IP and DNS information automatically* to *Use configured IP and DNS information* in order to lock in the IP addresses.

2. In some cases, the *System IP Settings* area may be blank because your router has not delivered the IP addresses. If so, enter the following IP addresses from your LAN administrator:

   a) Enter a static IP address for each TalkSwitch unit in the *Unit IP address* boxes.

   b) Enter the *Subnet mask* for the LAN. This address determines the subnet the unit IP addresses belong to.

   c) Enter the IP address of the *Default gateway* on your network. A gateway is a hardware device that connects the office network to the Internet. The router may act as default gateway.

   d) Enter the IP address of the *Preferred DNS server*. DNS is a service used to resolve a domain name to an IP address. The router may act as DNS server.

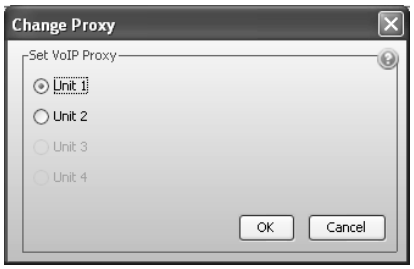
   e) If applicable, enter the IP address of the *Alternate DNS server*.

TalkSwitch proxy

If your TalkSwitch system has multiple TalkSwitch units, the *TalkSwitch Proxy* area displays and allows you to change the unit acting as local proxy. The local proxy handles Internet communications for the TalkSwitch system. It establishes connections across the Internet, and then routes Internet data to and from each TalkSwitch unit. The lowest numbered TalkSwitch unit acts as local proxy by default.

The router must map the SIP, TFTP and HTTP ports to the local proxy. See *Router port forwarding* on page 36.

1. If necessary, click the **Change Proxy** button. The **Change Proxy** window appears.



2. Select the TalkSwitch unit.

Public IP address

The *Public IP address* section allows you to set up Internet parameters so the TalkSwitch system can communicate with other locations over the Internet.

1. Set the *Type of public address*. Choices are:
   - *Dynamic public IP address* — This is the default setting. Your ISP (Internet Service Provider) will assign different public IP addresses to your location. The TalkSwitch system will check its public IP address every five minutes. When the public IP address changes, the TalkSwitch system will automatically use the new one. This allows it to manage VoIP calls properly.

     If *Dynamic public IP address* is selected, the window completes the *Current public IP address*, and sets *Public IP address-checker server name* to **checkip.talkswitch.com**. This server runs an IP checker application that provides public IP addresses to TalkSwitch systems. You cannot change these items.

   - *Static public IP address* — A static IP address is fixed. Your ISP assigns the static IP address. If this location will have the SIP server for a TalkSwitch VoIP network, we recommend using a static IP address.

     If *Static public IP address* is selected, the window allows you to enter the *Current public IP address*, and disables the *Public IP address-checker server name* box.

2. If you selected *Dynamic public IP address*, enter the *Fully qualified domain name* (FQDN). Get the FQDN from your ISP. The FQDN is only required if this location is the SIP server for a TalkSwitch VoIP network, or has external IP extensions.

   A DDNS (Dynamic Domain Name Service) provider such as **www.dyndns.com** matches your dynamic IP addresses to your FQDN, so your TalkSwitch VoIP network or external IP phones will continue to work when the IP address changes.

   If you are using DDNS, ensure your router supports your DDNS service, and configure it to update the DNS servers.

If your router does not support DDNS, download one of the applications specified on **www.dyndns.com**. To update the DNS servers, the application needs to run on a PC connected to the same LAN as the TalkSwitch system.

3. If you selected *Static public IP address*, enter the *Current public IP address* from your ISP. Leave the *Fully qualified domain name* box blank.

   If the TalkSwitch unit is not behind a router, or if a private virtual network is used, the public IP address should be set to the local IP address of the TalkSwitch unit acting as local proxy.

   Note that it will take up to one minute for the new static public IP address to take effect.

Router port forwarding

The *Router Port Forwarding* area displays the type of gateway device (i.e. the type of router), the IP address of the gateway (i.e. router), and whether router configuration is required.

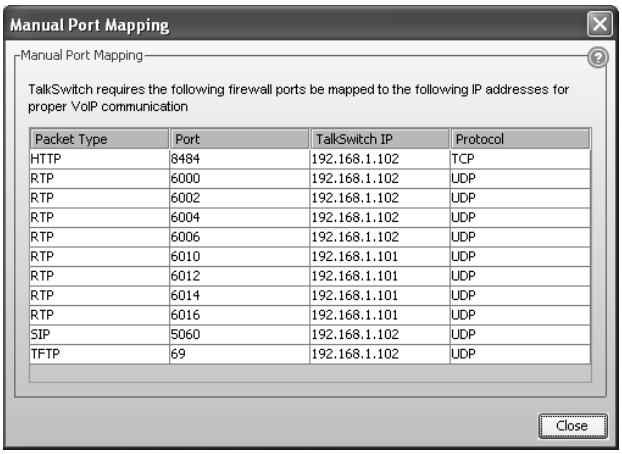
A router is a gateway between the local area network and the Internet. Most routers have a firewall to block unwanted data from the Internet. For voice data to reach the TalkSwitch system through the firewall, port forwarding is required. Port forwarding allows the router to map ports to the IP addresses of the TalkSwitch units. Valid Internet data will use the ports to go through the firewall to the TalkSwitch units.

If you are setting up external IP extensions, a TalkSwitch VoIP network, or a VoIP service that doesn't handle port forwarding, port forwarding is required.

If port forwarding is required, and your router supports uPNP (Universal Plug and Play), ensure uPNP is enabled. The TalkSwitch system will use uPNP to automatically set up port forwarding, and the **Automatic (uPNP Enabled)** link will appear. No router configuration is required.

If port forwarding is required but your router doesn't support uPNP, or automatic port forwarding doesn't work, the **Manual Port Mapping Required** link will appear. You will need to configure the router as described below.

1. If required, click the **Manual Port Mapping Required** link. The **Manual Port Mapping** window appears. It lists the packet type, port number, IP address and protocol of each required port.


Manual Port Mapping
Manual Port Mapping
TalkSwitch requires the following firewall ports be mapped to the following IP addresses for proper VoIP communication
Close

Packet Type	Port	TalkSwitch IP	Protocol
HTTP	8484	192.168.1.102	TCP
RTP	6000	192.168.1.102	UDP
RTP	6002	192.168.1.102	UDP
RTP	6004	192.168.1.102	UDP
RTP	6006	192.168.1.102	UDP
RTP	6010	192.168.1.101	UDP
RTP	6012	192.168.1.101	UDP
RTP	6014	192.168.1.101	UDP
RTP	6016	192.168.1.101	UDP
SIP	5060	192.168.1.102	UDP
TFTP	69	192.168.1.102	UDP

2. To access the router configuration:

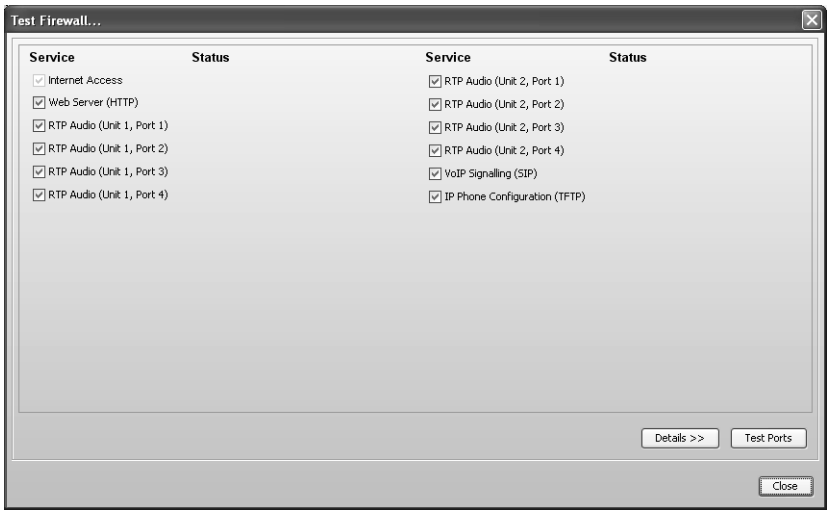
   a) Click the link containing the IP address of the gateway. The default browser starts, and prompts you for the router's user name and password.

   b) Enter the router's user name and password. The browser shows a setup screen.

   c) Navigate to the screen used to set up port forwarding. See your router documentation.

   d) Set up port forwarding using the information from the **Manual Port Mapping** window. See your router documentation for instructions on how to map ports. For information on configuring routers and mapping ports, visit **http://www.portforward.com/english/routers/port_forwarding/routerindex.htm**.

3. To check the status of each port through the firewall, click **Check Firewall**. The **Test Firewall** window appears.



4. Select the services you want to check.

5. Click **Test Ports**. The system will check the ports of the selected services.

Configuring the router manually

If you cannot access the router configuration through the **IP Configuration** page, configure the router manually.

1. Open the router configuration and navigate to the screen used to set up port forwarding. See your router documentation.

2. Map port 5060 (Type: UDP) to the TalkSwitch unit acting as local proxy. Port 5060 is the default port for forwarding SIP signaling data to the TalkSwitch system.

   If required, you can map a different port. In this case, enter the port in the *SIP signalling port* box of the **VoIP Settings** window.

3. Map the following ports (Type: UDP) to the TalkSwitch units. These RTP ports forward voice traffic to the TalkSwitch system by default.

VoIP Lines/Unit	Unit 1	Unit 2	Unit 3	Unit 4
4	6000–6006	6010–6016	6020–6026	6030–6036
8	6100–6114	6200–6214	6300–6314	6400–6414

If required, you can map different ports. In this case enter the first port in the *Starting RTP port* box of the **VoIP Settings** window.

4. If you are setting up external IP extensions, map port 69 (Type: UDP) to the TalkSwitch unit acting as local proxy. Port 69 allows the TalkSwitch system to configure external IP extensions.

5. If you are setting up TalkSwitch IP phones as external IP extensions, map port 8484 (Type: TCP) to the TalkSwitch unit acting as local proxy. Port 8484 allows the TalkSwitch system to configure TalkSwitch IP phones as external IP extensions.

6. Map port 9393 (Type: TCP) to the TalkSwitch unit acting as local proxy. Port 9393 allows remote configuration of the TalkSwitch system.

7. If available, enable Quality of Service (QoS) to give voice traffic priority over data.

8. Save the configuration to the router.

ON-HOLD/RINGBACK

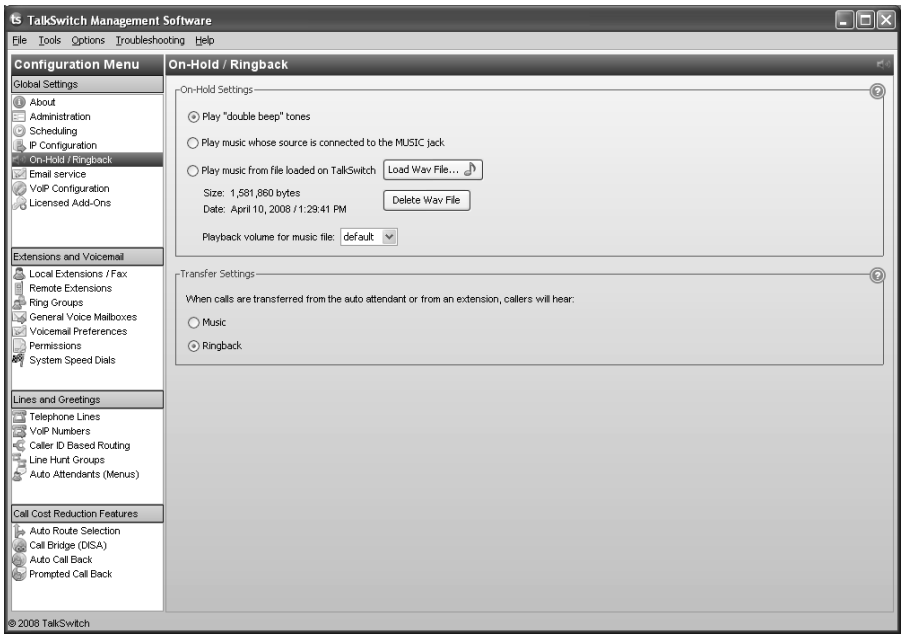
Music on hold plays when a caller is on hold, or is being transferred to an extension. The system plays a double beep tone or ringback tone by default, but you can have it play music or a recorded message instead.

The TalkSwitch system can play music on hold from an internal .wav file or from an external audio source. An external audio source is a CD player or MP3 player. It must be connected to the MUSIC jack on the back of each TalkSwitch unit. See *Setting up music on hold* on page 15.

The recording time for internal music on hold, voicemail, and the auto attendants is shared on the TalkSwitch unit. You can expand the built-in memory using memory cards.

The **On-Hold/Ringback** page allows you to set up music on hold.

1. Select the **On-Hold/Ringback** page.



On-hold settings

The *On-hold Settings* area allows you to select the sound to play while the caller is on hold.

1. Select the sound to play while the caller is on hold. Choices are:

   - *Play "double beep" tones* — Plays a "beep beep" sound.
   - *Play music whose source is connected to the MUSIC jack* — Plays music from a CD player connected to the MUSIC jack. See *Setting up music on hold* on page 15.
   - *Play music from file loaded on TalkSwitch* — Plays the .wav file loaded into the TalkSwitch system.

2. If you selected *Play music from file loaded on TalkSwitch*:

   a) Obtain an 8 kHz, 8 bit, Mono, u-Law .wav file that contains the music or recorded message. You can download free music on hold from the mytalkswitch section of the TalkSwitch website.

   You can also convert .mp3 and other file formats to .wav with a utility like Audio Converter and Ripper from **www.audioi.com**.

   b) Click **Load Wav file**. The **Select Wav File** window appears.

   c) Click **Browse** to select a .wav file, and then click **Select Wav File**. The system will load the .wav file into the TalkSwitch unit(s), and will display the file size and date loaded.

3. Adjust the volume of music on hold.

   a) Check the volume by placing a test call and going on hold.

   b) Change the volume by setting the *Playback volume for music file* list. Choices are:

      - Loudest
      - Louder
      - Default
      - Quieter
      - Quietest

   c) Choose **File > Save**.

   You must save the configuration for the volume setting to take effect.

   d) Repeat Steps a to c.

Transfer settings

The *Transfer Settings* area allows you to select the sound to play while the caller is being transferred from an auto attendant or to another extension.

1. Select the sound to play while the caller is being transferred. Choices are:

    - *Music* — Plays music on hold as configured in the *On-Hold Settings* area.
    - *Ringback* — Plays the ringback tone, which is the normal sound heard when the other person's phone is ringing.

Deleting a music on hold file

You can delete a music on hold file. This may be required if you need more space for voicemail messages. In this case you can use an external audio source for music on hold, or you can upgrade the memory capacity of the TalkSwitch units. See *Increasing memory capacity* on page 19.

1. Click **Delete Wav File**. The system deletes the .wav file from the TalkSwitch unit(s).

Playing music on hold on the PA system

You can play music on hold through the PA jack.

1. Dial ***80** to play or stop playing music on hold through the PA jack.

If music on hold is playing through the PA jack, and a user dials ***0** to make an overhead page, the overhead page will interrupt the music on hold. However, if the PA jack is configured for voicemail screening, and music on hold is playing, voicemail screening will not interrupt the music on hold. See *PA Output* on page 183.

EMAIL SERVICE

The TalkSwitch system can send an e-mail if a voicemail message has been left in a mailbox. The e-mail includes the caller ID, and can include the voicemail message as an attachment.

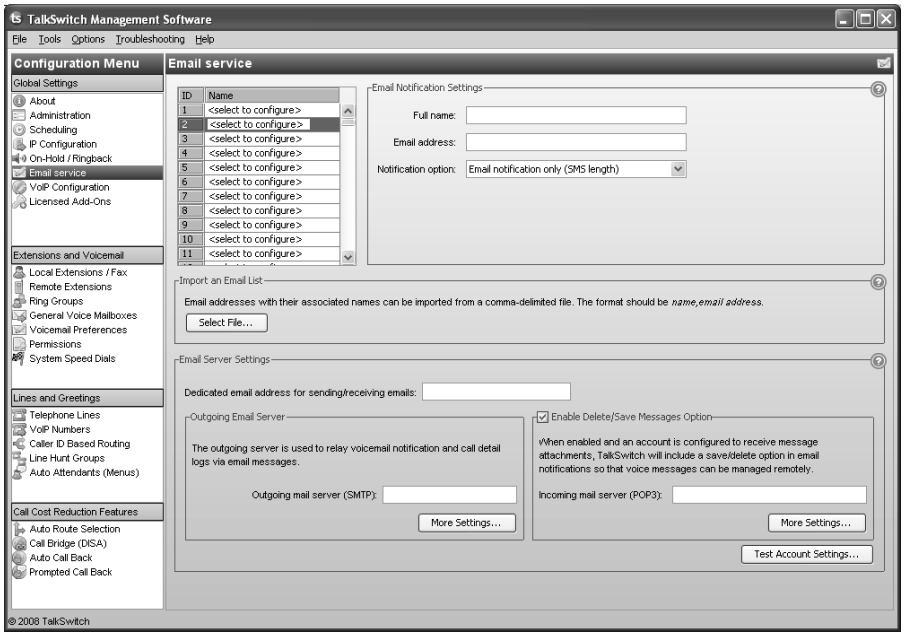
The e-mail can optionally include links to delete or save the voicemail message. If you delete the voicemail message, it will be removed from the TalkSwitch system. If you save the voicemail message, the system will change its status from "new" to "saved". A voicemail message that is saved will no longer activate the new message indicator on the user's extension.

The system can send notification to up to four e-mail addresses per mailbox. For example, if a caller leaves a message in the general mailbox for the sales group, the system can send an e-mail to four members of the sales group.

An e-mail address can be assigned to multiple mailboxes. For example, the system can send e-mails to a user if callers leave messages in the user's local extension mailbox, remote extension mailbox or a general mailbox.

The **Email Service** page allows you to maintain e-mail addresses, set up the e-mail server parameters, and test the e-mail server.

1. Select the **Email Service** page.



Email notification settings

The *Email Notification Settings* area allows you to add up to 255 e-mail addresses.

1. Select a slot for the e-mail address.
2. Enter the recipient's *Full name*.
3. Enter the recipient's *Email address*.
4. Select the *Notification option*. Choices include:

   - *Include voice message as attachment* — Attaches the voicemail message to the e-mail as a .wav file.

     ```
     From: TalkSwitch Voicemail
     Sent: April 18, 2008 2:53:30 PM
     Subject: From: ext 112
     Attachments: voice_message_111.wav (68 KB)
     _____________________________________

     Voice Mailbox 111

     New message from ext 112
     Received: Friday, April 18 at 2:52pm
     Length: 6 seconds

     Total un-reviewed messages: 5
     Total saved messages: 0
     _____________________________________
     ```

```
TalkSwitch Voicemail Management

Save message      Delete message
_____________     _______________
Note: It may take up to 2 minutes for the message to be saved/
deleted from your mailbox.
```

- *Email notification only (full length)* — Does not attach the voicemail message to the e-mail, but includes full details about the contents of the voice mailbox.

```
From: TalkSwitch Voicemail
Sent: April 18, 2008 2:53:30 PM
Subject: From: ext 112
_____________________________________
Voice Mailbox 111

New message from ext 112
Received: Friday, April 18 at 2:52pm
Length: 6 seconds

Total un-reviewed messages: 5
Total saved messages: 0
_____________________________________
```

- *Email notification only (SMS length)* — Does not attach the voicemail message to the e-mail, and sends minimum details about the message. The notification is less than 130 characters, so is short enough to be sent in one e-mail to a cell phone.

```
From: TSvm
Sent: April 18, 2008 2:53:30 PM
Subject: New msg

Mbox 111
Msg from ext 112
Rcvd: Fri, Apr 18, 2:52pm
Lngth: 6s
New: 5
Saved: 0
```

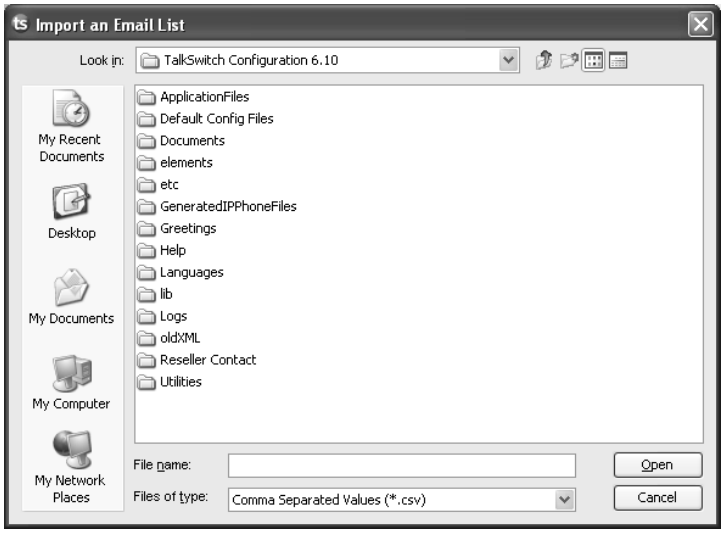
Import an email list

The *Import an Email List* area allows you to import e-mail addresses from a .csv file.

Each entry in the file must be formatted as follows: name,e-mail address. For example:

```
John Doe,john.doe@email.com
Jane Doe,jane.doe@email.com
```

1. Click **Select File**. The **Import an Email List** window appears.



2. Select the .csv file, and then click **Open**.

E-mail server settings

The *Email Server Settings* area allows you to set up the e-mail server parameters, include links to delete or save the voicemail message, and test the system's ability to send e-mail.

You will need an e-mail address to send notifications from. For best results, use an e-mail associated with an SMTP server that supports Plain authentication.

Notification may not work properly if the SMTP server uses Microsoft Exchange Server, and/or certain security features are enabled.

Use e-mail account information from your Internet service provider (ISP), system administrator, or alternatively, obtain one from a compatible online e-mail provider such as **Lavabit.com**.

1. Enter an e-mail address in the *Dedicated email address for sending/receiving emails* box. When e-mail notifications arrive from the system, this e-mail address will show up in the *From* field.

2. Enter the name of the *Outgoing mail server (SMTP)*.

3. To configure authentication and server port numbers, click **More Settings**. The **Internet Email Settings** window appears. See *Outgoing server authentication* on page 44 and *Outgoing server port numbers* on page 45.

4. To include links within the e-mail notification, select the *Enable delete/save messages option* checkbox. The links will allow the recipient to delete or save the voicemail message. The e-mail notification can only include the links if the voicemail message is attached.

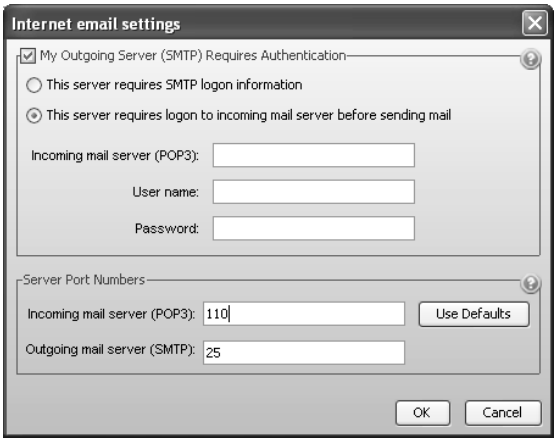
Deleting the voicemail message will remove it from the TalkSwitch unit. Saving the voicemail message will change its status from "new" to "saved". A voicemail message that is saved will no longer activate the new message indicator on the user's extension.

5. Enter the name of the *Incoming mail server (POP3)*. The name will automatically appear in the *My Outgoing Server (SMTP) Requires Authentication* area.

6. To configure authentication, server port numbers and how often the TalkSwitch unit should check the POP3 server, click **More Settings**. The **Internet email settings** window appears. See *Incoming server authentication* on page 45, *Incoming server port numbers* on page 45 and *Incoming mail server options* on page 45.

Outgoing server authentication

The *My Outgoing Server (SMTP) Requires Authentication* area allows you to enter account information for the outgoing server. The outgoing server is also referred to as the SMTP server. It can have its own account information, or can use the account information from the incoming server. The incoming server is also referred to as the POP3 server.



1. If the outgoing server requires authentication, select the *My Outgoing Server (SMTP) Requires Authentication* checkbox.

2. Choose the type of authentication used by the outgoing server:

   - If the outgoing server uses its own logon information, and can send e-mail without first logging on to the incoming server:

     i) Select *This server requires SMTP logon information*.

     ii) Enter the *User name* of the account from the outgoing server.

     iii) Enter the *Password* of the account from the outgoing server.

   - If the outgoing server uses logon information from the incoming server, and must log on to the incoming server before sending e-mail:

     i) Select *This server requires logon to incoming mail server before sending mail.*

     ii) Enter the name of the *Incoming mail server (POP3)*. The name will automatically appear in the *Enable Delete/Save Messages Option* area.

     iii) Enter the *User name* of the account from the incoming server. The user name will automatically appear in the *Incoming Server Authentication* area.

iv) Enter the *Password* of the account from the incoming server. The password will automatically appear in the *Incoming Server Authentication* area.

Outgoing server port numbers

The *Server Port Numbers* area allows you to enter a port number for the outgoing server. This is only required if the standard default port number cannot be used.

1. The default *Outgoing mail server (SMTP)* port is 25. If required, enter a different port number ranging from 0 to 65535.

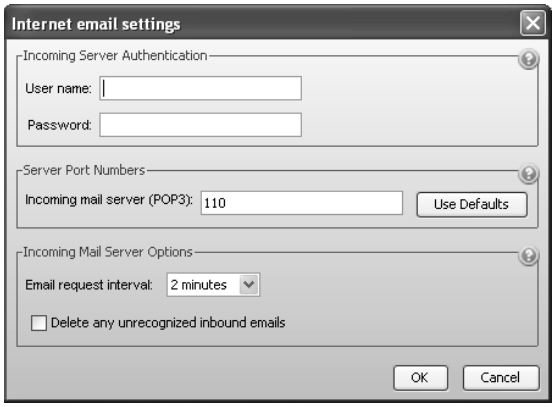
Incoming server authentication

The *Incoming Server Authentication* area allows you to enter the user name and password of the account from the incoming server. The incoming server is also referred to as the POP3 server.

1. Enter the *User name* of the account from the incoming server. The user name will automatically appear in the *My Outgoing Server (SMTP) Requires Authentication* area.
2. Enter the *Password* of the account from the incoming server. The password will automatically appear in the *My Outgoing Server (SMTP) Requires Authentication* area.

Incoming server port numbers

The *Server Port Numbers* area allows you to enter a port number for the incoming server. This is only required if the standard default port number cannot be used.



1. The default *Incoming mail server (POP3)* port is 110. If required, enter a different port number ranging from 0 to 65535.

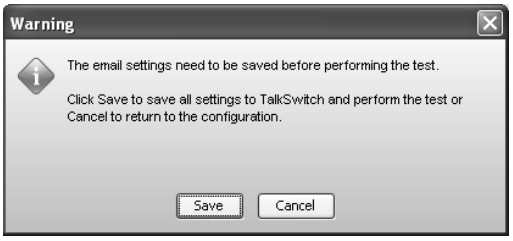
Incoming mail server options

The *Incoming Mail Server Options* area allows you to specify how often the TalkSwitch unit should check the incoming server for e-mail messages from users deleting and saving voicemail messages. The incoming server is also referred to as the POP3 server.

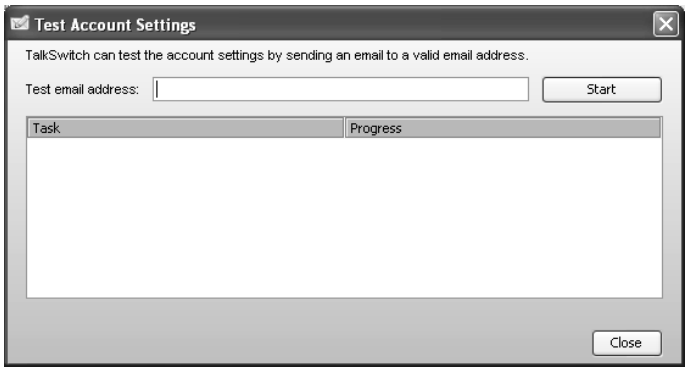
1. Select the *Email request interval*, ranging from *1 minute* to *30 minutes*.
2. To delete e-mail messages that aren't from TalkSwitch users (i.e. spam), select the *Delete any unrecognized inbound emails* checkbox.

Testing the e-mail server settings

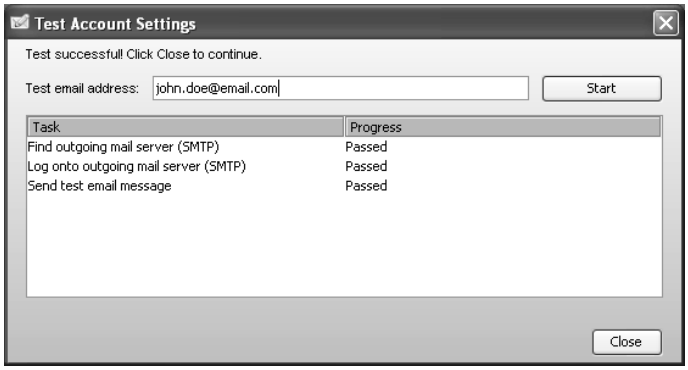
1. Click **Test Account Settings**. A window appears, prompting you to save the configuration.


Warning
The email settings need to be saved before performing the test.
Click Save to save all settings to TalkSwitch and perform the test or Cancel to return to the configuration.
Save
Cancel

2. Click **Save**. The system saves the configuration, then the **Test Account Settings** window appears.



3. Enter an e-mail address in the *Test email address* box, and then click **Start**. The following window shows a successful result, with each task completed.



   If a task fails, the TalkSwitch system is not able to send e-mail messages. Adjust the e-mail server parameters accordingly:

   - If *Find outgoing mail server (SMTP)* fails, check the SMTP server name in the *Outgoing mail server (SMTP)* box.
   - If *Log onto outgoing mail server (SMTP)* fails, check the SMTP server authentication parameters in the **Internet Email Settings** window.

4. Ensure the e-mail address has received the test e-mail. Note that the e-mail may have been routed to a junk or spam folder.

Managing voicemail messages

If the *Notification option* is set to *Include voice message as attachment*, the recipient can play, save or delete the voicemail message.

1. To play the voicemail message, the recipient double-clicks the attachment. The default .wav player opens the voicemail message.
2. To save the voicemail message, the recipient clicks **Save message**. The e-mail program creates a new e-mail message with the *To* and *Subject* fields completed. The recipient sends this e-mail message. Upon receiving the e-mail message, the TalkSwitch system will change the status of the voicemail message from "new" to "saved", and the local extension will turn off the new message indicator. However the system will not delete the voicemail message.

   Note that the system allows up to 99 voicemail messages per mailbox. Once a mailbox fills up, callers won't be able to leave voicemail messages for that user. Therefore users should delete voicemail messages before the mailbox fills up.
3. To delete the voicemail message, the recipient clicks **Delete message**. The e-mail program creates a new e-mail message with the *To* and *Subject* fields completed. The recipient sends this e-mail message. Upon receiving the e-mail message, the TalkSwitch unit will delete the voicemail message.

Setting up POP3 service with Microsoft Exchange

If your system uses Microsoft Exchange, and you wish to include voicemail messages as attachments within the notification emails, you must set up POP3 service.

For instructions on how to download, install and configure POP3 service, refer to the Microsoft documentation included with your version of Exchange. See **http://technet.microsoft.com/en-us/library/aa998454(EXCHG.65).aspx** for more information.

Risks of enabling POP3 service

- The default configuration does not use encryption when sending the username and password.
- Unauthorized clients can use the POP3 service.
- E-mails from users for saving and deleting voicemail messages are not stored on the Exchange server, but are pulled through to the client (i.e. the TalkSwitch system).

Security recommendations

1. Install the TalkSwitch system on the same LAN as the Exchange server.
2. Follow the Microsoft procedures for tightening security.
3. The POP3 service is a virtual server. You can use the following techniques to control incoming access to a virtual server. Refer to the Microsoft documentation included with your version of Exchange.
   - Grant or deny access using IP addresses or Internet domain names.
   - Require authentication for incoming connections.
   - Restrict concurrent connections.
   - Set connection time-out values.

VOIP CONFIGURATION

The **VoIP Configuration** page allows you to set up the TalkSwitch profile for the SIP server or a SIP client, and up to four service provider profiles. You can also view registrar entries, view registration status, set codec options, and reserve VoIP lines for specific setups.

The **VoIP Configuration** page is only present with a VoIP-enabled TalkSwitch unit. The layout of the page depends on whether you are configuring a TalkSwitch profile or a service provider profile.

Setting up a TalkSwitch profile

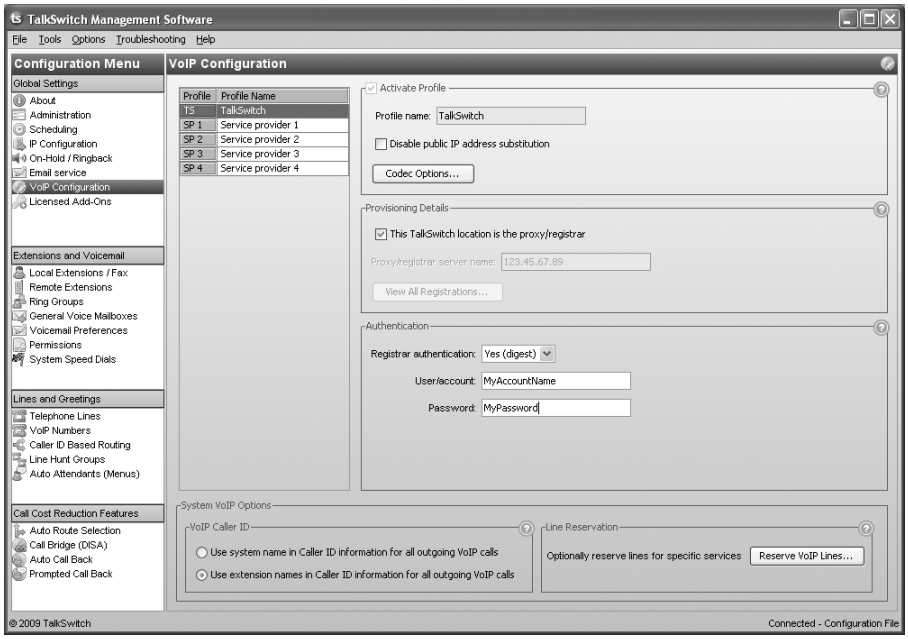
You can set up a TalkSwitch profile for the SIP server or for a SIP client. The TalkSwitch profile contains registration details and authentication information. The information is from the TalkSwitch unit acting as the SIP server for a TalkSwitch VoIP network.

Any VoIP-enabled TalkSwitch unit can act as the SIP server. However a TalkSwitch VoIP network requires that one (and only one) TalkSwitch unit be configured to act as the SIP server. TalkSwitch systems at other locations are configured as SIP clients.

When the SIP server is enabled and SIP clients are registered with the SIP server, all VoIP calls within the TalkSwitch VoIP network contact the SIP server to resolve the destination location. Once the destination is resolved, the call goes directly from the source to the destination.

Activate profile

1. Select the **VoIP Configuration** page.
2. Select **Profile TS**, which is for configuring the TalkSwitch profile.



The *Activate Profile* area indicates the *Profile name* is *TalkSwitch*.

Leave the *Disable public IP address substitution* checkbox cleared.

You can specify which codecs to use by clicking the **Codec Options** button. See *Setting codec options* on page 55.

Provisioning details

TalkSwitch SIP server

1. Select the *This TalkSwitch location is the proxy/registrar* checkbox. The page enables the **View Registrar Entries** button. See *Viewing registrar entries* on page 53.
2. If the SIP server requires a non-default SIP server registration interval, SIP signalling port, or starting RTP port, enter the new values. See *VoIP Trunking* on page 188.
3. Ensure your router is set up to perform port forwarding for the SIP signalling and RTP ports. See *Router port forwarding* on page 36.

TalkSwitch SIP client

1. Leave the *This TalkSwitch location is the proxy/registrar* checkbox cleared.

   The page disables the *Registrar authentication* list and the **View Registrar Entries** button.
2. Enter the *Current public IP address* or *Fully qualified domain name* (FQDN) of the SIP server into the *Proxy/registrar server name* box.

   If the SIP server requires a non-default SIP signalling port (i.e. if the SIP server cannot use 5060 as the SIP signalling port), specify the port number after a colon at the end of the public IP address or FQDN (e.g. 123.45.67.89:3456).
3. Ensure your router is set up to perform port forwarding for the SIP signalling port. See *Router port forwarding* on page 36.

The *Provisioning Details* area also allows you to view the status of registered SIP devices. See *Viewing registration status* on page 54.

Authentication

The *Authentication* area allows you to set a username and password for the TalkSwitch VoIP network if you choose to permit access exclusively to authorized devices.

1. If you are configuring the SIP server location, set *Registrar authentication* to *Yes (digest)*.
2. Enter the *User/account name*. All SIP clients must use this name.
3. Enter the *Password*. All SIP clients must use this password.

System VoIP options

VoIP Caller ID

The *VoIP Caller ID* area allows you to set up the source for the caller ID name for outbound VoIP calls. The same setting is used for the TalkSwitch profile and all service provider profiles. Extension names are used by default.

1. Set the caller ID information for outgoing VoIP calls:
   - To use the *System name* from the **Administration** page, select *Use system name in caller ID information for all outgoing VoIP calls*.
   - To use the *First name* and *Last name* from the **Local Extensions/Fax** page, select *Use extension names in caller ID information for all outgoing VoIP calls*.

See *Caller ID settings* on page 65 to select the phone number that will appear on the other phone.

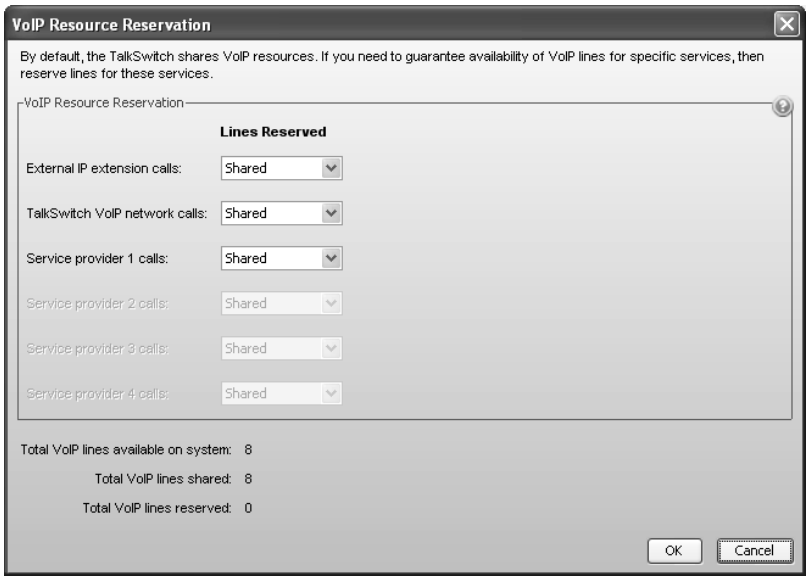
Line reservation

By default, all TalkSwitch VoIP lines are available for external IP extensions, TalkSwitch VoIP network calls, and/or service provider calls on a first-come first-served basis. You can also reserve VoIP lines for a specific use. For example, you could set aside two lines for external IP extensions.

Reserving VoIP lines guarantees resources for a specific network. Sharing VoIP lines uses them most efficiently.

When reserving VoIP lines for a service, the choice ranges from 1 to the number of unreserved VoIP lines at this location. *Shared* means do not reserve any VoIP lines for this service. Only unreserved lines will be available.

1. Click **Reserve VoIP Lines**. The **VoIP Resource Reservation** window appears.



2. Select the number of VoIP lines to reserve for *External IP extension calls* using its pull-down menu.
3. Select the number of VoIP lines to reserve for *TalkSwitch VoIP network calls* using its pull-down menu.
4. Select the number of VoIP lines to reserve for *VoIP service provider calls* using its pull-down menu.

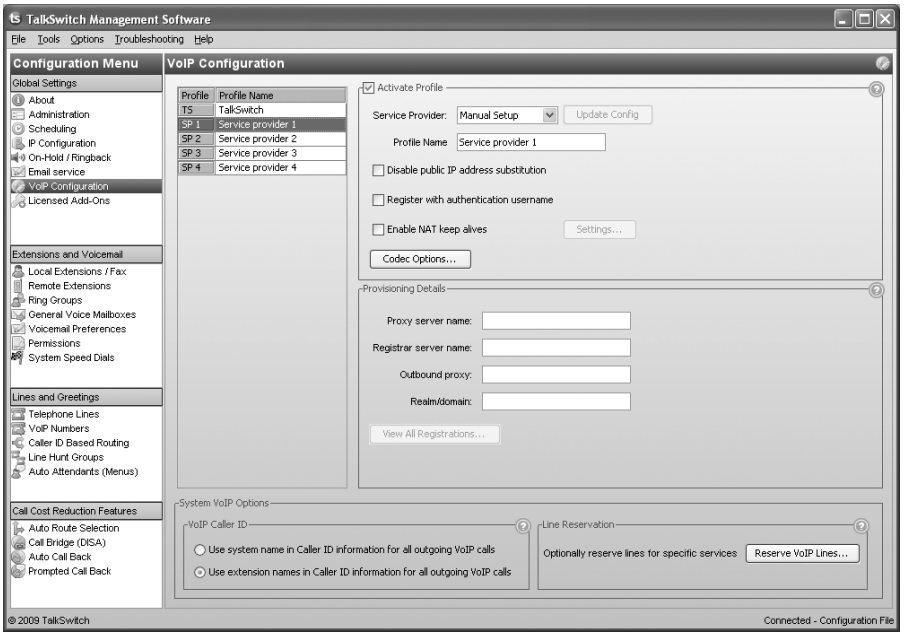
Setting up a service provider profile

A service provider profile contains the settings that allow your TalkSwitch to register with the provider.

If you are using a TalkSwitch-certified VoIP service provider, visit the support section at **http://global.talkswitch.com** to access the service configuration guide for your VoIP service provider.

1. Select the **VoIP Configuration** page.

2. Select a **Profile** (**SP 1** to **SP 4**).



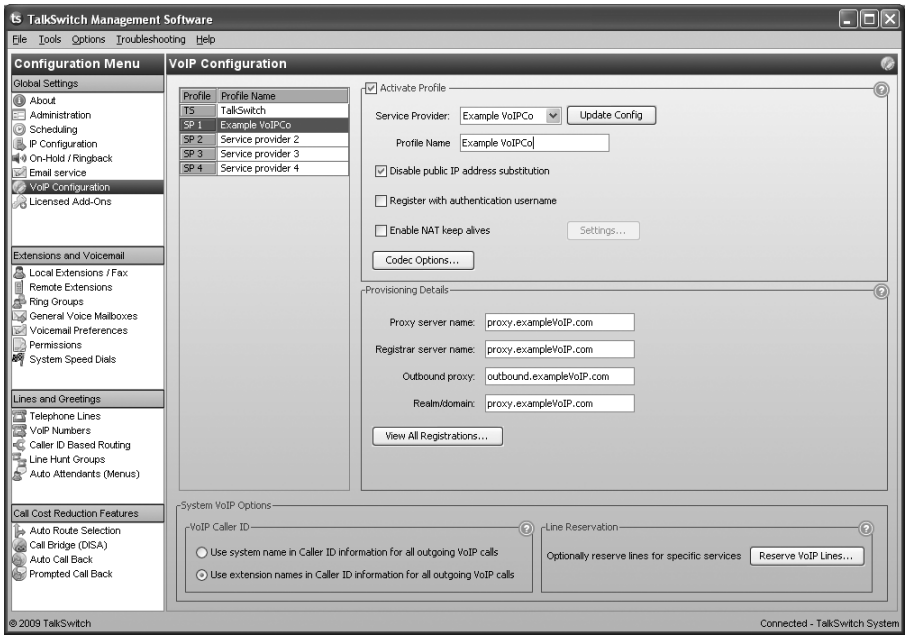
Activate profile

You can set up a service provider profile automatically or manually.

Automatic configuration

1. Select the *Activate Profile* checkbox.
2. The *Service Provider* menu offers a list of TalkSwitch certified VoIP service providers. If your service provider appears in the menu, click on the name. The name is then displayed in the *Service Provider* field.

3. Click the **Update configuration** button. The essential settings for communication with the service provider's registration servier will be completed automatically.



4. If you want to customize other aspects of your VoIP lines, you may do so in the *System VoIP Options* area. See *System VoIP options* on page 49.

   Account-specific and number-specific settings are not automatically configured. These must be entered on the **VoIP Numbers** page.

Manual configuration

1. Select the *Activate Profile* checkbox.
2. Enter the *Profile name*. The default profile name is *Service provider n* (e.g. *Service provider 1*).
3. If your service provider requires you to register using your private IP address, select the *Disable public IP address substitution* checkbox. Check with your service provider.
4. Some providers require the TalkSwitch system to register using the username or account information rather than the VoIP number(s) provided. If so, check the *Register with authentication username* box to have TalkSwitch register with the username information provided in the **VoIP numbers** page. Check with your VoIP service provider if you're uncertain which method of registration is required
5. If your service provider requires keep alive messages, and if your router does not support uPNP, check the *Enable NAT keep alives* checkbox.

   a) Click **Settings**. The **Keep Alive Settings** window appears.

   b) Select the method used to keep ports open. Choices are:

- *Simple ping* — A standard ping message that works with all SIP servers.
- *Nortel ping* — A ping message that works with Nortel SIP servers (e.g. Nortel MCS 5200).

c) If necessary, change the ping frequency. The default setting is 45 seconds.

6. You can specify which codecs to use by clicking the **Codec options** button. See *Setting codec options* on page 55.

Provisioning details

Enter the IP addresses or FQDNs, as provided by the service provider, into the following boxes. If the service provider does not specify a value, leave the box blank.

- *Proxy server name*
- *Registrar server name*
- *Outbound proxy*
- *Realm/domain*

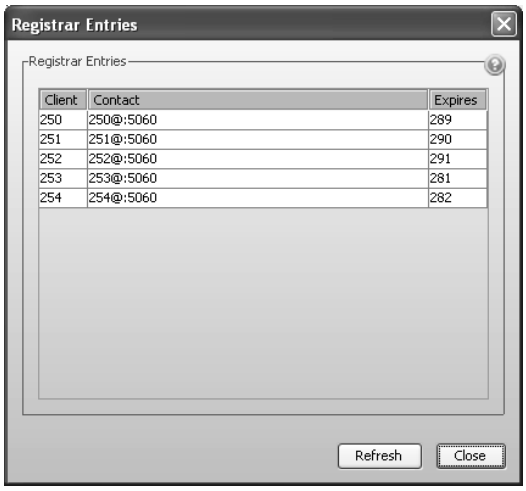
The **View All Registration** button will allow you to confirm connection to your service provider once you have completed the configuration of your VoIP numbers. See *Viewing registration status* on page 54.

If you want to customize other aspects of your VoIP lines, you may do so in the *System VoIP Options* area. See *System VoIP options* on page 49.

Viewing registrar entries

If you are configuring the TalkSwitch profile for the SIP server, the window enables the **View Registrar Entries** button. Clicking the button shows the **Registrar Entries** window with a list of the VoIP numbers, their IP addresses and port numbers, and the number of seconds until their registrations with the SIP server will expire.

1. Click **View Registrar Entries**. The **Registrar Entries** window appears.

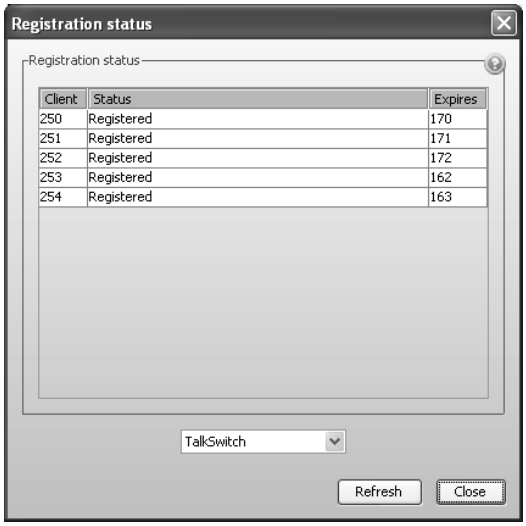


- *Client* lists the VoIP numbers set up within the TalkSwitch units.
- *Contact* lists the address information used by the SIP server to contact the client. It is comprised of the VoIP number, public IP address and port number.
- *Expires* is the amount of time, in seconds, until the client has to re-register with the SIP server.

Viewing registration status

Clicking the **View All Registrations** button shows a window with a list of VoIP numbers, their registration status, and the number of seconds until their registrations with the SIP server will expire. This confirms the TalkSwitch system is registered with a SIP server.

1. Click **View All Registrations**. The **Registration status** window appears.



2. Choose *All Registered Numbers* or an active profile.

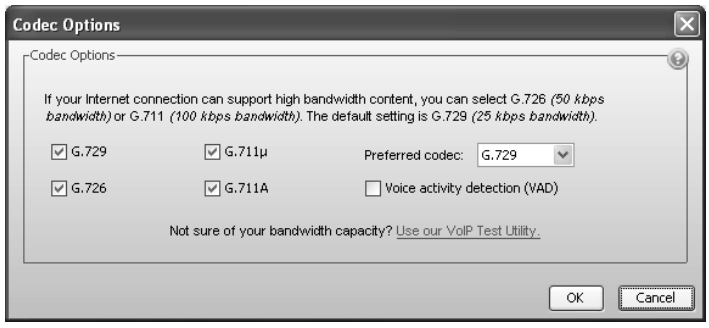
   - *Client* lists the VoIP numbers set up within the TalkSwitch units.
   - *Status* is the registration status (*Registered* or *Not registered*).
   - *Expires* is the amount of time, in seconds, until the client has to re-register with the SIP server.

Setting codec options

A codec is a method of compressing and decompressing audio signals for communication across a network. TalkSwitch supports the G.729, G.726 and G.711 (µ-law or A-law) codecs for VoIP calls. If your service provider or equipment requires specific codecs for VoIP or Fax over IP calls, you can restrict TalkSwitch to use the required codec.

The **Codec Options** window allows you to select the codecs your system can use, specify the preferred codec, and clear the unsupported codecs. You can specify the codecs for the TalkSwitch profile, and for each service provider profile. External IP extensions will use the preferred codec specified in the TalkSwitch profile.

1. Click **Codec Options**. The **Codec Options** window appears.


Codec Options
Codec Options
If your Internet connection can support high bandwidth content, you can select G.726 (50 kbps bandwidth) or G.711 (100 kbps bandwidth). The default setting is G.729 (25 kbps bandwidth).
G.729
G.711µ
Preferred codec:
G.729
G.726
G.711A
Voice activity detection (VAD)
Not sure of your bandwidth capacity? Use our VoIP Test Utility.
OK
Cancel

2. Select the codecs your system can use, and clear the unsupported codecs. The following codecs are supported:

   - *G.729* — This codec provides good quality. It requires the least bandwidth and accommodates the highest number of concurrent calls.
   - *G.726* — This codec provides better quality. It requires more bandwidth and accommodates fewer concurrent calls.
   - *G.711µ* — This codec provides high quality and supports Fax over IP. It requires the most bandwidth and accommodates the fewest number of concurrent calls. G.711µ is used in North America and Japan.
   - *G.711A* — This codec provides high quality and supports Fax over IP. It requires the most bandwidth and accommodates the fewest number of concurrent calls. G.711A is used worldwide outside North America and Japan.

3. Select the *Preferred codec*. *G.729* is the default codec, which uses the least bandwidth and accommodates the highest number of concurrent calls.

4. Set *Voice activity detection (VAD)*. Enabling VAD reduces voice bandwidth when no speech is detected, and reduces transmission of background noise. We recommend disabling VAD to keep bandwidth available for speech.

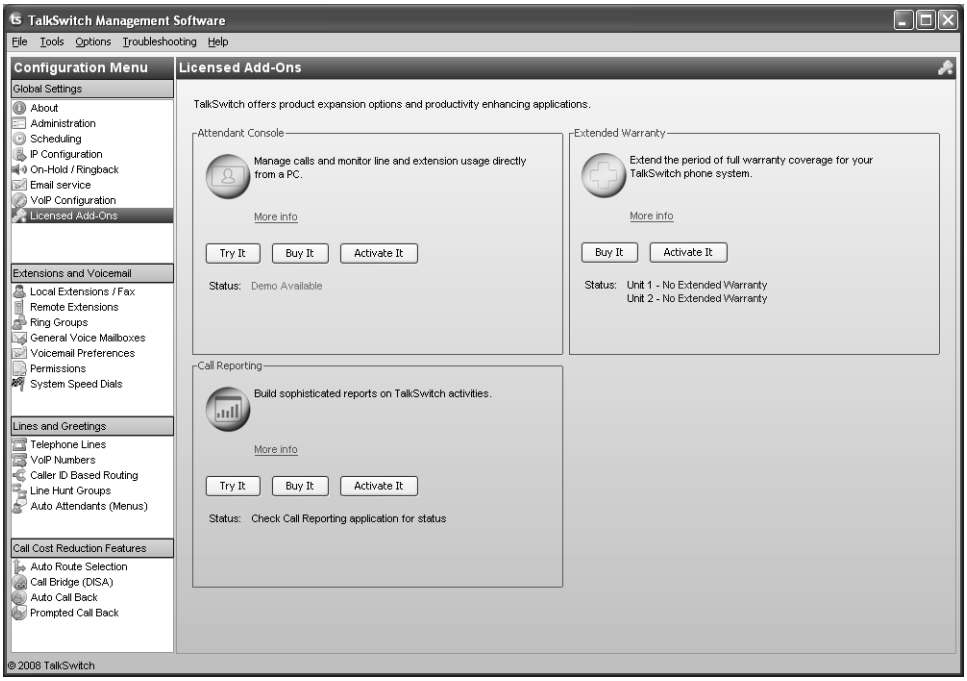
LICENSED ADD-ONS

Attendant Console allows users to monitor telephone lines, VoIP lines and extensions, and to perform call handling on a PC.

Call Reporting polls the call detail record (CDR) output from the TalkSwitch unit, and writes the information to a database file. It then reads the database file, and generates reports.

An extended warranty covering TalkSwitch units is available in the United States and Canada.

The **Licensed Add-Ons** page allows you to try, buy and activate Attendant Console and Call Reporting, and to buy and activate an extended warranty.



The **Buy It** button displays contact information according to the location of installation. Installers have the option to have their contact information display in this window by following these steps:

1. Copy the varinfo.html and logo.png files from the \Reseller Contact subdirectory to the TalkSwitch program directory (e.g. \TalkSwitch Configuration 6.12\).
2. Edit the copied version of varinfo.html to contain the desired contact information.
3. If you want to also display your logo, edit the copied logo.png file to contain your logo.

Your contact information will display provided these new files are in the TalkSwitch program directory.

LOCAL EXTENSIONS/FAX

A local extension can either be an analog extension or an IP extension.

An analog extension is a device (standard phone, cordless phone, fax machine or modem) connected to an extension jack on the TalkSwitch unit.

An IP extension can be internal or external. An internal IP extension is an IP phone connected through the LAN to the TalkSwitch unit. An external IP extension is an IP phone located outside the office. It is connected to the TalkSwitch system through the Internet. External IP extensions require a VoIP-enabled TalkSwitch unit. See *Adding IP phones* on page 81 for more information.

Each TalkSwitch unit can have up to 16 local extensions. Either four or eight local extensions can be analog or IP extensions, depending on the model of the TalkSwitch unit. The remaining twelve or eight local extensions can only be IP extensions.

All local extensions are active by default. If you plug a standard phone into an extension jack, it will be ready to make and receive calls. However configuration is required to set up an IP extension.

The first eight local extensions are numbered as follows:

- TalkSwitch Unit 1 — 111 to 118
- TalkSwitch Unit 2 — 121 to 128
- TalkSwitch Unit 3 — 131 to 138
- TalkSwitch Unit 4 — 141 to 148

The next eight local extensions can only be IP extensions, and are numbered as follows:

- TalkSwitch Unit 1 — 151 to 158
- TalkSwitch Unit 2 — 161 to 168
- TalkSwitch Unit 3 — 171 to 178
- TalkSwitch Unit 4 — 181 to 188

Note that dialing 101–108 will ring local extensions 111–118.

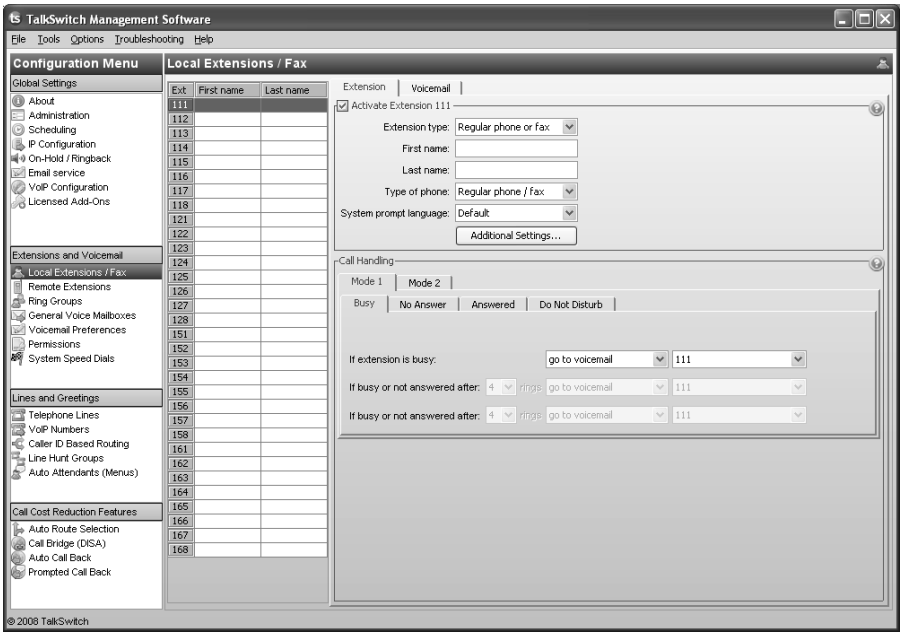
The **Local Extensions/Fax** page allows you to set up a local extension.

Extension tab

The *Extension* tab allows you to set up the extension type, user name, make and model, prompt language, IP extension details, programmable keys, direct line access, hotline, hunt group permissions, caller ID, and call cascade for each local extension.

1. Select the **Local Extensions/Fax** page.

2. Select the *Extension* tab.



Activate extension

The *Activate Extension* area allows you to enter information about the user and the extension.

If you connected a TalkSwitch TS-9112i, TS-9133i or TS-480i IP phone to the LAN and used it to select an extension number, the *Extension type* and *Type of phone* for that extension will be complete. In this case, you will enter the *First name* and *Last name*, and will select the *System prompt language*.

1. Select the extension number.
2. If necessary, select the *Activate Extension* checkbox. All extensions are enabled by default.
3. Select the type of extension in the *Extension type* list. Choices are *Regular phone or fax* and *IP extension*. *Regular phone or fax* allows you to set up an analog extension.
4. Enter the user's *First name* and *Last name*. The names are used for caller ID, and appear within the configuration pages. The *Last name* is used with the dial-by-name directory.
5. Select the *Type of phone*. The TalkSwitch system uses this to configure the phone with supported features like intercom paging, group paging and the message waiting counter.
   - If you set *Extension type* to *Regular phone or fax*, choices are *Regular phone/fax*, *TS-80*, *TS-100*, *TS-200*, *TS-400* and *TS-600*.
   - If you set *Extension type* to *IP extension*, you can choose the make and model. The models of IP phones that are available vary by market — see your reseller for details. Supported IP phones are:
     - TalkSwitch — TS-550i, TS-480i, TS-350i, TS-9133i, TS-9112i and TS-850i
     - Polycom — IP 301, IP 320, IP 430, IP 501, IP 550, IP 601 and IP 650
     - Grandstream — GXP 2000 and GXP 2020
     - Counterpath — eyeBeam softphone

We strongly recommend you only use supported IP phones. However, if you have an unsupported IP phone, you can select *Other IP phone*. Other IP phones that support the G.711 codec (µ-law or A-law) may work with the system, but some features may not work. As TalkSwitch cannot enable special features, or customization of these IP phones, further configuration will be limited to the programmable options on the phone itself.

6. Select the language for prompts heard by the user of the extension in the *System prompt language* list.

7. Click **Additional Settings** to set up:

   - Direct line access (see *Direct line access* on page 61)
   - Hotline access (see *Hotline access* on page 63)
   - Hunt group permissions (see *Hunt group permissions* on page 64)
   - Caller ID settings (see *Caller ID settings* on page 65)

IP extension details

If you set *Extension type* to *IP extension*, the *IP Extension Details* area is enabled. The layout of the area depends on the type of phone. It allows you to set the location and identity of the IP phone, the time zone for an external IP extension, the programmable keys of a TalkSwitch TS-550i, TS-480i, TS-350i or TS-9133i, and the handset ID of a TalkSwitch TS-850i.

Once you save the configuration and reboot the IP phone, the TalkSwitch system will configure the IP phone. *Status* will change to *Registered*, and the IP address of the phone will appear.

If you connected a TalkSwitch IP phone to the LAN and used it to select an extension number, the *Location* and *MAC address* for that extension will be complete. In this case, you can configure the programmable keys of a TalkSwitch TS-550i, TS-480i, TS350i or TS-9133i.

1. Set the *Location* of the phone. Choices are:

   - *Internal* — An internal IP extension is located within the office, and is connected to the same LAN as the TalkSwitch unit.
   - *External* — An external IP extension is located outside the office, and is connected to the TalkSwitch unit via Internet. A VoIP-enabled TalkSwitch unit is required.
   - *Both* — The phone can be used as an internal or external IP extension. A VoIP-enabled TalkSwitch unit is required. This is available for the Counterpath eyeBeam softphone only.

Warning: Using an external IP extension to call an emergency service number will not send the correct address to the emergency operator. We strongly recommend that you apply a warning label to any external IP extension stating:

If an emergency call is made from this phone, you must provide your address to the emergency operator.

2. If you set *Location* to *External*, select the *Time zone* that matches the location of the IP phone.

3. Use one of the procedures below to identify the IP phone, depending on the make and model.

TalkSwitch TS-550i, TS-480i, TS350i, TS-9133i or TS-9112i

1. Set the *MAC address* of the phone:

   - The *MAC address* will be complete, if you connected a phone to the LAN and used it to select the extension number. See *Connecting the TalkSwitch IP phone to the network* on page 85.

   - You can select the *MAC address* from a list of automatically-detected phones connected to your LAN. This method is also available if you programmed the phone as an external IP extension (see *Programming the TalkSwitch IP phone as an external IP extension* on page 94). To use this method:

     i) Click the **Select** button. A **MAC Selection** window appears and lists TalkSwitch IP phones of the selected type. If a phone has already been assigned to an extension, *Ext Assignment* shows the extension number, first name and last name. If a phone can be assigned to the extension, *Ext Assignment* shows *Available*.

     ii) Select the *MAC address* of the IP phone associated to the extension, and then click **Select**.

   - You can enter the *MAC address* manually. The MAC address is printed on the bottom of the phone and on the phone's shipping box.

2. If the *Type of phone* is *TalkSwitch TS-550i*, *TS-480i*, *TS-350i* or *TS-9133i*, click **Configure Keys** to set up the programmable keys. See *Programming TS-9133i function keys* on page 93, *Programming TS-480i softkeys* on page 94, *Programming TS-550i function keys* on page 88, or *Programming TS-350i function keys* on page 87.

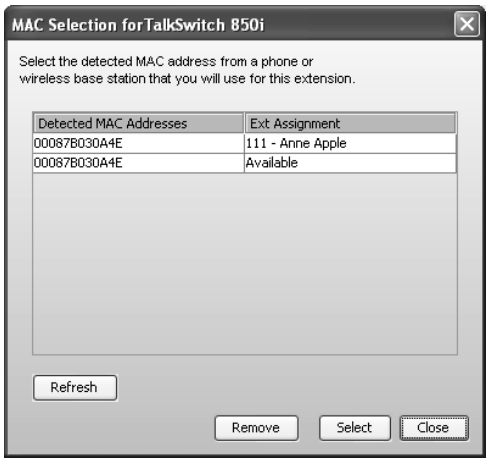
3. Choose **File > Save**. The system will create a configuration file that the phone will download when the phone is restarted.

TalkSwitch TS-850i

1. Set the *MAC address* of the phone.

   - You can select the *MAC address* from a list of automatically-detected bases connected to your LAN. To use the **MAC Selection** window:

     i) Click **Select**. A **MAC Selection** window appears. *Detected MAC addresses* lists the MAC addresses of bases connected to the LAN. A base can accommodate up to eight handsets. If a handset has already been assigned to an extension, *Ext Assignment* shows the extension number, first name and last name. If a handset can be assigned to the extension, *Ext Assignment* shows *Available*.



     ii) Select *Available*, and then click **Select**.

   - You can enter the *MAC address* manually. This is required if you are setting up an external IP extension. The MAC address is printed under the base associated with the handset.

2. Select the *Handset ID* for the extension. Use the handset name on the handset screen.

3. Choose **File > Save**. The system will create a configuration file that the phone will download when the phone is restarted.

Polycom or Grandstream

1. Enter the *MAC address*. The MAC address is printed on the bottom of the phone or on the phone's shipping box.

Counterpath or other IP phone

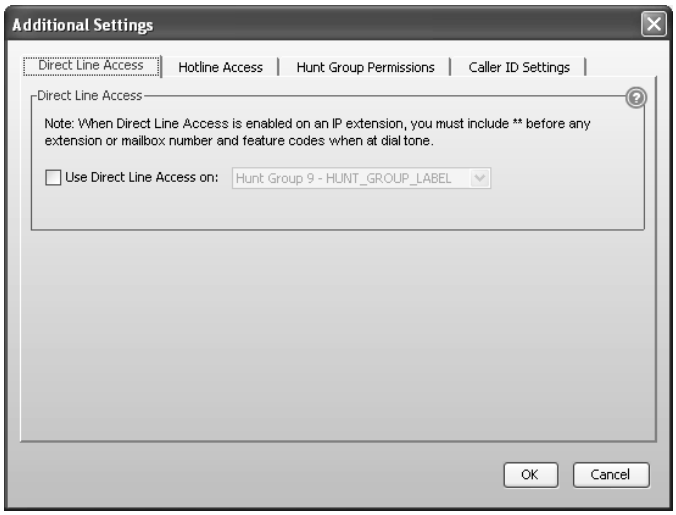
1. Enter the *Username* and *Password*. The default settings are user[extension number] and pass[extension number]. These are used to manage calls between TalkSwitch and the IP extension and must match the username and password of the IP phone. Change only if required.

Direct line access

The *Direct Line Access* area allows you to select the hunt group that the local extension will use automatically. When an analog phone goes off-hook or a number is dialed on an IP phone, the TalkSwitch unit will automatically find an available line within that hunt group.

Example: You have a fax machine connected to the local extension and don't want to reprogram the speed dial numbers with hunt group numbers. Enable direct line access and select the hunt group. As soon as the fax goes off-hook, it finds an available line within the line hunt group.

1. Click **Additional Settings**. The **Additional Settings** window appears.
2. Select the *Direct Line Access* tab.


Additional Settings
Direct Line Access
Hotline Access
Hunt Group Permissions
Caller ID Settings
Direct Line Access
Note: When Direct Line Access is enabled on an IP extension, you must include ** before any extension or mailbox number and feature codes when at dial tone.
Use Direct Line Access on:
Hunt Group 9 - HUNT_GROUP_LABEL
OK
Cancel

3. Select the *Use direct line access on* checkbox. The window enables the list of line hunt groups.
4. Select the hunt group.

Note: When using direct line access, you hear the telephone company dial tone when you pick up the handset. You do not hear the internal dial tone. The following features are only available from the internal dial tone:

- Intercom calls
- System speed dial numbers
- Calling the receptionist
- Attaching an account code
- Intercom paging
- Group paging
- Overhead paging
- Hunt groups
- Stutter dial tone for new voicemail
- Voicemail retrieval/access
- Entering command mode
- Call pickup
- Retrieving a parked call
- Do Not Disturb functions

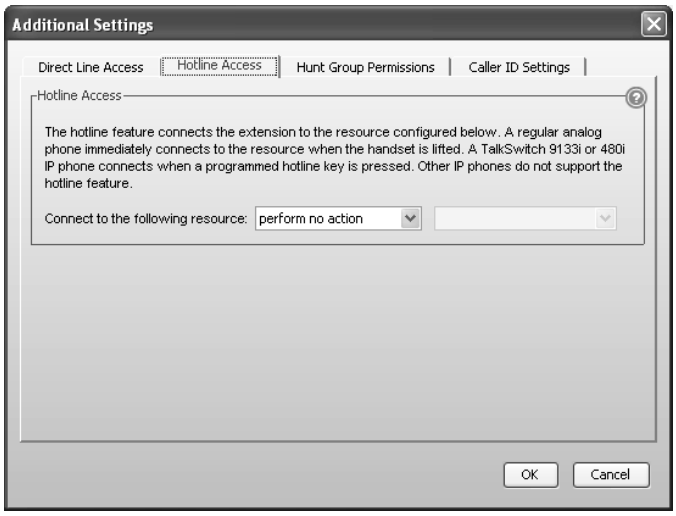
To obtain an internal dial tone on an analog extension that is set for direct line access, pick up the handset, and then press **Flash** or **Recall**. On an IP extension, press ******, and then dial the number or function.

Hotline access

The *Hotline Access* area allows you to select the resource that the extension will connect to. This restricts the extension to one special task, and you cannot use the extension for any other purpose.

An analog phone, TS-350i or TS-550i IP phone will automatically connect to the resource when you lift the handset, or press the speaker or headset button. A TS-9133i will connect to the resource when you press the top programmable function key. A TS-480i will connect to the resource when you press the **HotLine** softkey. Other IP phones do not support hotline access.

1. Click **Additional Settings**. The **Additional Settings** window appears.
2. Select the *Hotline Access* tab.

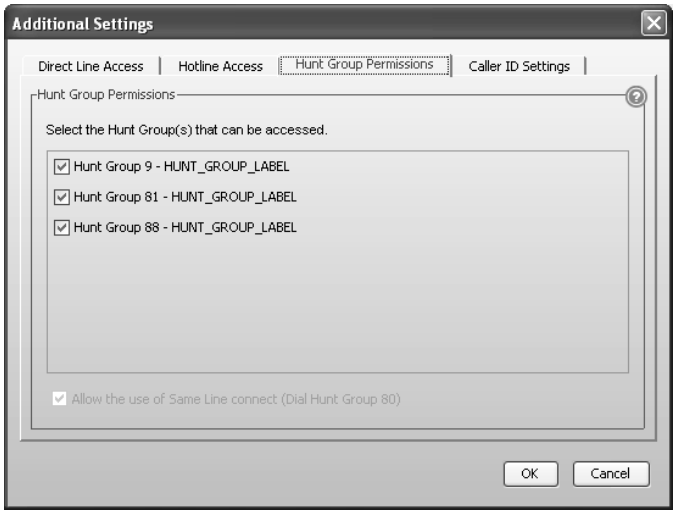

Additional Settings
Direct Line Access
Hotline Access
Hunt Group Permissions
Caller ID Settings
Hotline Access
The hotline feature connects the extension to the resource configured below. A regular analog phone immediately connects to the resource when the handset is lifted. A TalkSwitch 9133i or 480i IP phone connects when a programmed hotline key is pressed. Other IP phones do not support the hotline feature.
Connect to the following resource:
perform no action
OK
Cancel

3. Select the action in the *Connect to the following resource* list. Choices are:
   - *perform no action* — Does not automatically connect to a resource.
   - *go to voicemail* — Connects to the selected voicemail.
   - *go to local extension* — Connects to the selected local extension.
   - *go to remote extension* — Connects to the selected remote extension.
   - *go to ring group* — Connects to the selected ring group.
   - *go to auto attendant* — Connects to the selected auto attendant.
   - *go to VoIP location* — Connects to the selected VoIP number within the TalkSwitch VoIP network.
4. Select the resource. Depending on the action, resources are voice mailboxes, extensions, auto attendants, or VoIP numbers within the TalkSwitch VoIP network.

Hunt group permissions

The *Hunt Group Permissions* area allows you to select the hunt groups that the extension can use. You can restrict an extension's access to specific hunt groups. Each extension has access to all hunt groups by default.

1. Click **Additional Settings**. The **Additional Settings** window appears.
2. Select the *Hunt Group Permissions* tab.

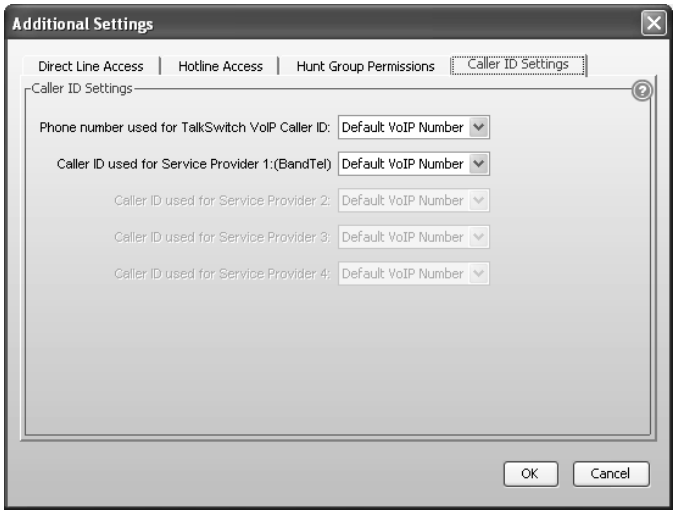

Additional Settings
Direct Line Access
Hotline Access
Hunt Group Permissions
Caller ID Settings
Hunt Group Permissions
Select the Hunt Group(s) that can be accessed.
Hunt Group 9 - HUNT_GROUP_LABEL
Hunt Group 81 - HUNT_GROUP_LABEL
Hunt Group 88 - HUNT_GROUP_LABEL
Allow the use of Same Line connect (Dial Hunt Group 80)
OK
Cancel

3. Set the checkboxes in the *Hunt Group Permissions* area to control access to the hunt groups. Select a checkbox to grant access to the hunt group, or clear the checkbox to remove access.
4. If *3-Way Calling/Conference* is selected in the **Telephone Lines** page, you can select the *Allow the use of same line connect (by dialing 80)* checkbox. If this is selected, the user can transfer a call to a remote extension using the same telephone line that the caller is on. The user dials **Flash** or **Recall**, remote extension, and then **Flash 4** or **Recall 4**. The telephone company puts the caller on hold, and the TalkSwitch system uses the same telephone line to try the remote extension.

Caller ID settings

The *Caller ID Settings* area allows you to select the phone number that will appear on the other phone when the local extension is used to make a VoIP call.

1. Click **Additional Settings**. The **Additional Settings** window appears.
2. Select the *Caller ID Settings* tab.



3. Set the *Phone number used for TalkSwitch VoIP caller ID* list to the phone number that should be displayed during a call on the TalkSwitch VoIP network.
4. Set each *Caller ID used for service provider* list to the phone number that should be displayed during a call through the VoIP service provider.

Choices include *Default VoIP number*, and the VoIP numbers set up in the **VoIP Numbers** page. If *Default VoIP number* is selected, the phone number associated with the VoIP line is used. This selection makes the most sense if your VoIP numbers have been configured to handle all inbound calls the same way.

See *System VoIP options* on page 49 to select the name that will appear on the other phone.

About call cascades

The *Call Handling* area allows you to set up call cascades for the local extension. A call cascade routes an incoming call, depending upon whether the local extension is busy, unanswered, answered, or in Do Not Disturb mode. You can set up different call cascades for Mode 1 and Mode 2. When the system is in Holiday Mode, all the local extensions will either use their Mode 1 call cascades or their Mode 2 call cascades.

For example, if a call reaches your desk and you are away, the call can be transferred to another local extension. If that extension is not answered, the call can be routed to your cell phone. If you do not answer your cell phone, the call can be transferred to your voice mailbox.

Note that the system can be configured to ignore the call cascade for an unanswered intercom call, or for an unanswered transferred call.

How call cascades work

A call interacts with an extension's call cascade in three phases: the call enters the call cascade, is routed by the call cascade, and then exits the call cascade.

Entering the call cascade

First, a call enters the extension's call cascade. This occurs when:

- The caller dials the extension.
- The caller selects an auto attendant option that transfers the call to the extension.
- The caller leaves a message in a voice mailbox or listens to an announcement, and then presses the **0** option (**9** in some regions) to route the call to an extension.
- A user manually transfers the call to the extension.

Being routed by the call cascade

Second, the call cascade routes the call. It can try up to three routing alternatives. If the extension is not available, the call cascade tries the first alternative. If that is not available, it tries the second. If the second alternative is not available, it goes to the third. To route the call, the call cascade can:

- Attempt to transfer the call to a local extension, remote extension or ring group. Note that the other extension's call cascade is ignored.
- Attempt to transfer the call to a VoIP number within the TalkSwitch VoIP network.
- Transfer the call to a voice mailbox, allowing the caller to leave a message. The call can be transferred to a local extension, remote extension or general voice mailbox. The system hangs up after the caller leaves a message.
- Play an announcement. The system hangs up after playing the announcement.
- Route the call to an auto attendant.
- Transfer the call to the call queue if the local extension or ring group is busy. The call is placed on hold. The TalkSwitch system will ring the local extension when it becomes available.
- Play a call waiting tone on the user's phone if the extension is busy.
- Play a busy tone on the caller's phone if the extension is busy.
- Keep ringing the extension.
- Hang up.

Exiting the call cascade

Third, the call exits the call cascade. This occurs when:

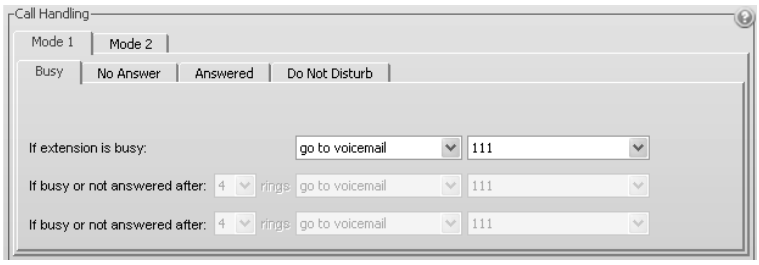
- A user answers the call, or accepts it as a screened transfer.
- The call cascade transfers the call to a voice mailbox or announcement, and the caller presses the **0** option (**9** in some regions) to route the call to an extension, voice mailbox, announcement or auto attendant.
- The call cascade routes the call to an auto attendant.
- The call cascade transfers the call to the call queue of a local extension or ring group.

Setting up call handling

Busy call cascade

The busy call cascade is used when the extension is busy.

1. Select the *Mode 1* tab or the *Mode 2* tab.
2. Select the *Busy* tab.



3. Set up the first alternative.

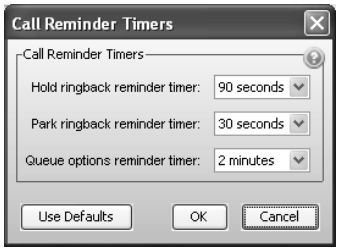
   a) Select the action in the *If extension is busy* list. Choices are:

   - *go to voicemail* — Transfers the call to the selected voicemail.
   - *go to local extension* — Attempts to transfer the call to the selected local extension.
   - *go to remote extension* — Attempts to transfer the call to the selected remote extension.
   - *go to ring group* — Attempts to transfer the call to the selected ring group.
   - *play announcement* — Plays the selected announcement.
   - *invoke call waiting* — Notifies the user that a caller is attempting to reach them.
   - *go to auto attendant* — Routes the call to the selected auto attendant.
   - *queue at extension* — Transfers the call to the local extension's call queue.
   - *play busy tone* — Plays a busy tone in the caller's phone.
   - *hang up* — Disconnects the telephone line.
   - *go to VoIP location* — Attempts to transfer the call to the selected VoIP number within the TalkSwitch VoIP network.

   b) Select the resource. Depending on the action, resources are voice mailboxes, extensions, announcements, auto attendants, or VoIP numbers within the TalkSwitch VoIP network.

   c) If you selected *queue at extension*, the caller will hear a prompt each time the reminder timer expires. The prompt says *"If you wish to continue holding, please remain on the line. To leave a voicemail message, press 1"*. You can set the duration of the reminder timer, to control how often the caller hears the prompt. See *Call Reminders on page 187*.

i) Choose **Options > Call Reminders**. The **Call Reminder Timers** window appears.



ii) Set the *Queue options reminder timer*.

4. If you selected an extension in the first alternative, set up the second alternative.

   a) Set the *If busy or not answered after* list to the number of rings to try the first alternative. Choices range from 1 to 9.

   b) Set the action. Choices are:

      - *go to voicemail*
      - *go to local extension*
      - *go to remote extension*
      - *go to ring group*
      - *play announcement*
      - *go to auto attendant*
      - *hang up*
      - *go to VoIP location*

   c) Select the resource. Depending on the action, resources are voice mailboxes, extensions, announcements, auto attendants, or VoIP numbers within the TalkSwitch VoIP network.

5. If you selected an extension in the second alternative, set up the third alternative.

   a) Set the *If busy or not answered* list to the number of rings to try the second alternative. Choices range from 1 to 9.

   b) Set the action. Choices are:

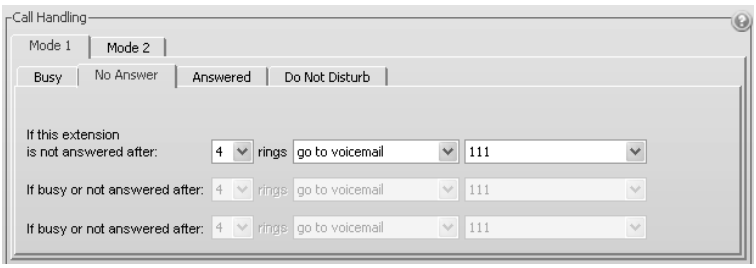
      - *go to voicemail*
      - *go to auto attendant*
      - *hang up*
      - *play announcement*

   c) Select the resource. Depending on the action, resources are voice mailboxes, announcements, or auto attendants.

No answer call cascade

The no answer call cascade is used when the extension is not answered.

1. Select the *Mode 1* tab or the *Mode 2* tab.
2. Select the *No Answer* tab.



3. Set up the first alternative.
   a) Set the *If this extension is not answered after* list to the number of rings to try the extension. Choices range from 1 to 9.
   b) Select the action. Choices are:
      - *go to voicemail*
      - *go to local extension*
      - *go to remote extension*
      - *go to ring group*
      - *play announcement*
      - *go to auto attendant*
      - *keep ringing*
      - *hang up*
      - *go to VoIP location*
   c) Select the resource. Depending on the action, resources are voice mailboxes, extensions, announcements, auto attendants, or VoIP numbers within the TalkSwitch VoIP network.
4. If you selected an extension in the first alternative, set up the second alternative.
   a) Set the *If busy or not answered after* list to the number of rings to try the first alternative. Choices range from 1 to 9.
   b) Set the action. Choices are:
      - *go to voicemail*
      - *go to local extension*
      - *go to remote extension*
      - *go to ring group*
      - *play announcement*

- *go to auto attendant*
- *hang up*
- *go to VoIP location*

c) Select the resource. Depending on the action, resources are voice mailboxes, extensions, announcements, auto attendants, or VoIP numbers within the TalkSwitch VoIP network.

5. If you selected an extension in the second alternative, set up the third alternative.

a) Set the *If busy or not answered after* list to the number of rings to try the second alternative. Choices range from 1 to 9.

b) Set the action. Choices are:

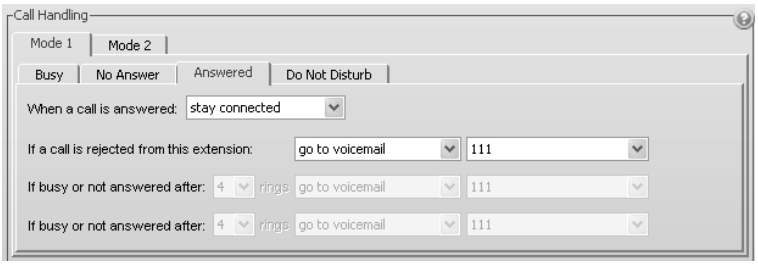
- *go to voicemail*
- *go to auto attendant*
- *hang up*
- *play announcement*

c) Select the resource. Depending on the action, resources are voice mailboxes, announcements, or auto attendants.

Answered call cascade

The answered call cascade is used when the extension is answered.

1. Select the *Mode 1* tab or the *Mode 2* tab.
2. Select the *Answered* tab.



3. Set the *When a call is answered* list. Choices are:

- *stay connected* — Transfers the call to the extension. This option disables the remaining controls in the tab.
- *play caller's name first* — Performs a screened transfer. The caller is prompted to state their name. When the user answers the phone, the system plays the caller's name, and then prompts the user to accept the call. If the user accepts the call by pressing **#**, it is routed to the extension. If the user rejects the call by pressing ***** or hanging up, the call is routed to the first alternative.

4. Set up the first alternative.

a) Set the *If a call is rejected from this extension* list to the action. Choices are:

- *go to voicemail*

- *go to local extension*
- *go to remote extension*
- *go to ring group*
- *play announcement*
- *go to auto attendant*
- *hang up*
- *go to VoIP location*

b) Select the resource. Depending on the action, resources are voice mailboxes, extensions, announcements, auto attendants, or VoIP numbers within the TalkSwitch VoIP network.

5. If you selected an extension in the first alternative, set up the second alternative.

a) Set the *If busy or not answered after* list to the number of rings to try the first alternative. Choices range from 1 to 9.

b) Set the action. Choices are:

- *go to voicemail*
- *go to local extension*
- *go to remote extension*
- *go to ring group*
- *play announcement*
- *go to auto attendant*
- *hang up*
- *go to VoIP location*

c) Select the resource. Depending on the action, resources are voice mailboxes, extensions, announcements, auto attendants, or VoIP numbers within the TalkSwitch VoIP network.

6. If you selected an extension in the second alternative, set up the third alternative.

a) Set the *If busy or not answered after* list to the number of rings to try the second alternative. Choices range from 1 to 9.

b) Set the action. Choices are:

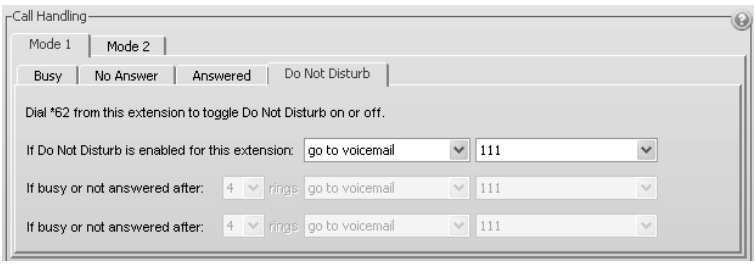
- *go to voicemail*
- *go to auto attendant*
- *hang up*
- *play announcement*

c) Select the resource. Depending on the action, resources are voice mailboxes, announcements, or auto attendants.

Do not disturb cascade

The do not disturb call cascade is used when the extension is in Do Not Disturb mode.

1. Select the *Mode 1* tab or the *Mode 2* tab.
2. Select the *Do Not Disturb* tab.



3. Set up the first alternative.
   a) Set the *If do not disturb is on for this extension* list. Choices are:
      - *go to voicemail*
      - *go to local extension*
      - *go to remote extension*
      - *go to ring group*
      - *play announcement*
      - *go to auto attendant*
      - *hang up*
      - *go to VoIP location*
   b) Select the resource. Depending on the action, resources are voice mailboxes, extensions, announcements, auto attendants, or VoIP numbers within the TalkSwitch VoIP network.
4. If you selected an extension in the first alternative, set up the second alternative.
   a) Set the *If busy or not answered after* list to the number of rings to try the first alternative. Choices range from 1 to 9.
   b) Set the action. Choices are:
      - *go to voicemail*
      - *go to local extension*
      - *go to remote extension*
      - *go to ring group*
      - *play announcement*
      - *go to auto attendant*
      - *hang up*
      - *go to VoIP location*

c) Select the resource. Depending on the action, resources are voice mailboxes, extensions, announcements, auto attendants, or VoIP numbers within the TalkSwitch VoIP network.

5. If you selected an extension in the second alternative, set up the third alternative.

   a) Set the *If busy or not answered after* list to the number of rings to try the second alternative. Choices range from 1 to 9.

   b) Set the action. Choices are:

      - *go to voicemail*
      - *go to auto attendant*
      - *hang up*
      - *play announcement*

   c) Select the resource. Depending on the action, resources are voice mailboxes, announcements, or auto attendants.

Setting do not disturb mode

The user can toggle Do Not Disturb mode by dialing ***62** on their local extension.

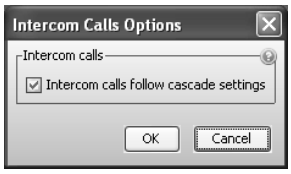
The administrator can enable or disable Do Not Disturb mode for a local extension by phone.

1. Pick up a local extension, or remotely call the TalkSwitch system.
2. Enter command mode by either pressing **#** on an analog extension phone, or ***55#** on a TalkSwitch IP phone (note: other brands may use ***55 Send** or ***55 Dial**).
3. Enter the system password, followed by **#**.
4. Enter one of the following commands:
   - [Local extension] **60#** to disable Do Not Disturb mode for the local extension.
   - [Local extension] **61#** to enable Do Not Disturb mode for the local extension.
5. Press ***** to exit command mode.

Ignoring call cascades for unanswered intercom calls

An intercom call is an internal call from a local extension to another local extension, remote extension or ring group. If the call is unanswered, the call follows that extension's call cascade by default. However you can configure the system to ignore call cascades for unanswered intercom calls. See *Internal Calls* on page 189.

1. Choose **Options > Internal Calls**. The **Intercom Calls Options** window appears.



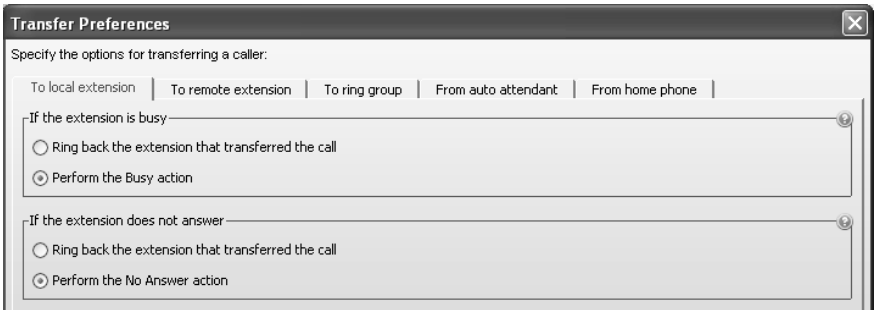
2. To have the system ignore call cascades for unanswered intercom calls, clear the *Intercom calls follow cascade settings* checkbox.
   - If the extension is busy, the caller will hear a busy tone.

- If the extension is not answered, it will continue to ring.
- If a local extension has call screening enabled, the caller won't be prompted for their name. However, if a remote extension or ring group has call screening enabled, call screening will occur. If the call is rejected, the system will follow the call cascade.
- If a local extension has do not disturb mode enabled, it won't ring, and the caller will hear *"I'm sorry, that extension is unavailable at this time. Do not disturb on. Please try again later"*. If all the local extensions in a ring group have do not disturb mode enabled, the caller will hear a busy tone.

Ignoring call cascades for calls transferred from local extensions

A user at a local extension can transfer a call to another local extension, remote extension or ring group. If the call is unanswered, the call follows that extension's call cascade by default. However you can configure the local extensions, remote extensions and/or ring groups to ignore their call cascades for calls that are transferred from a local extension but unanswered. In this case, the system will ring back the local extension that transferred the call, reconnecting the user with the caller they transferred.

1. Choose **Options > Transfer Preferences**. The **Transfer Preferences** window appears.


Transfer Preferences
Specify the options for transferring a caller:
To local extension
To remote extension
To ring group
From auto attendant
From home phone
If the extension is busy
Ring back the extension that transferred the call
Perform the Busy action
If the extension does not answer
Ring back the extension that transferred the call
Perform the No Answer action

2. Select the appropriate tab.
   - To configure call handling for calls transferred to local extensions, select the *To Local Extension* tab (see *To local extension tab* on page 184).
   - To configure call handling for calls transferred to remote extensions, select the *To Remote Extension* tab (see *To remote extension tab* on page 184).
   - To configure call handling for calls transferred to ring groups, select the *To Ring Group* tab (see *To ring group tab* on page 186).
3. Select the call handling setup for calls transferred from a local extension but unanswered:
   - To have the system ring back the local extension that transferred the call, select *Ring back the extension that transferred the call*.
   - To use the Busy call cascade, select *Perform the Busy action*.
   - To use the No Answer call cascade, select *Perform the No Answer action*.

Voicemail tab

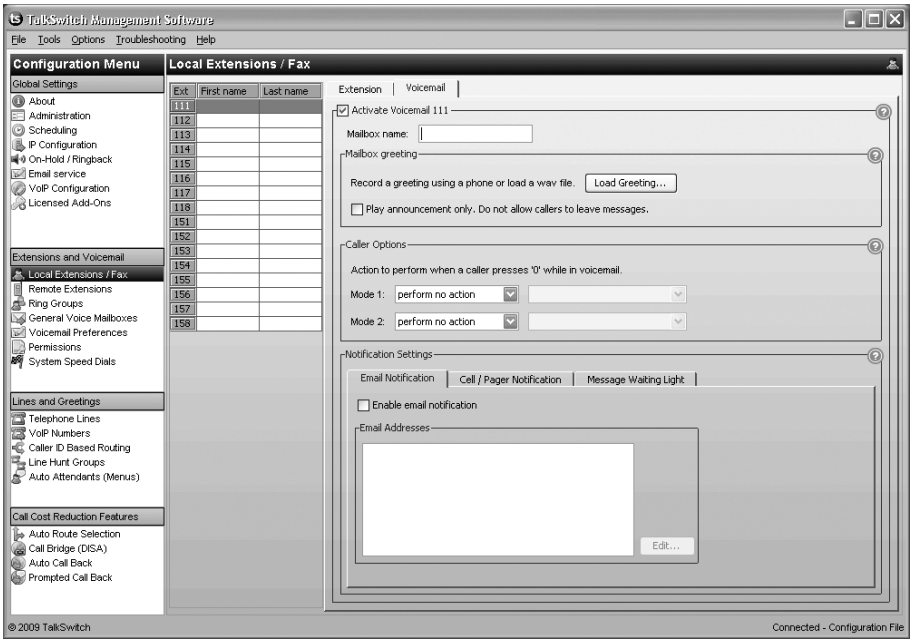
The *Voicemail* tab allows you to activate the voice mailbox or announcement, load the greeting, set the action to perform if the caller dials **0** (**9** in some regions), and set up voicemail notification.

By default, each local extension mailbox is activated.

The recording time for internal music on hold, voicemail, and the auto attendants is shared on the TalkSwitch unit. You can expand the built-in memory using memory cards.

Note that the system allows up to 99 voicemail messages per mailbox. Once a mailbox fills up, callers won't be able to leave voicemail messages for that user. Therefore users should delete voicemail messages before the mailbox fills up.

1. Select the **Local Extensions/Fax** page.
2. Select the *Voicemail* tab.



Note: Depending on the region, an operator may be dialed using 9 or 0.

3. If necessary, select the *Activate Voicemail* checkbox.

Mailbox greeting

The *Mailbox Greeting* area allows you to load a greeting, and configure the voice mailbox as a mailbox or announcement. Note that you can also record a greeting using a local extension connected to the TalkSwitch unit.

The greeting should tell the caller to dial **0** (**9** in some regions) to perform the action selected in *Caller options*, if you will configure these options.

1. To record a greeting:

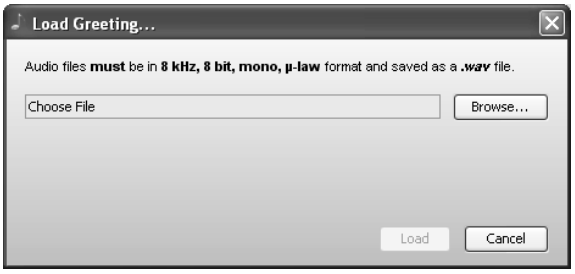
   a) Pick up a local extension connected to the TalkSwitch unit.

   b) Press ** <extension number>, and then follow the prompts to record a greeting.

2. To load a greeting:

   a) Obtain an 8 kHz, 8 bit, Mono, u-Law .wav file that contains the greeting.

   b) Click **Load Greeting**. The **Load Greeting** window appears.



   c) Click **Browse** to select a .wav file, click **Load Greeting**, and then click **Load**. The system will load the .wav file into the TalkSwitch unit.

3. To configure the voice mailbox as a mailbox, leave the *Play announcement only* checkbox cleared. The TalkSwitch unit activates the greeting and the mailbox, and enables all the controls on the page. The caller will hear a greeting, and will be able to leave a message in the mailbox.

4. To configure the voice mailbox as an announcement, select the *Play announcement only* checkbox. The TalkSwitch unit activates the greeting, disables the ability to leave a message, and disables the *Notification Settings* area. The caller will hear a greeting, but will not be able to leave a message.

Caller options

The *Caller Options* area sets the action to perform if the caller dials **0** (**9** in some regions) during the voicemail greeting. You can set up different actions for Mode 1 and Mode 2.

1. For each mode, select the action to perform if a caller dials **0** (**9** in some regions). Choices are:

   - *go to voicemail*
   - *go to local extension*
   - *go to remote extension*
   - *go to ring group*
   - *play announcement*

- *go to auto attendant*
- *perform no action*

2. Depending on the action, enter the extension number or auto attendant.

Notification settings

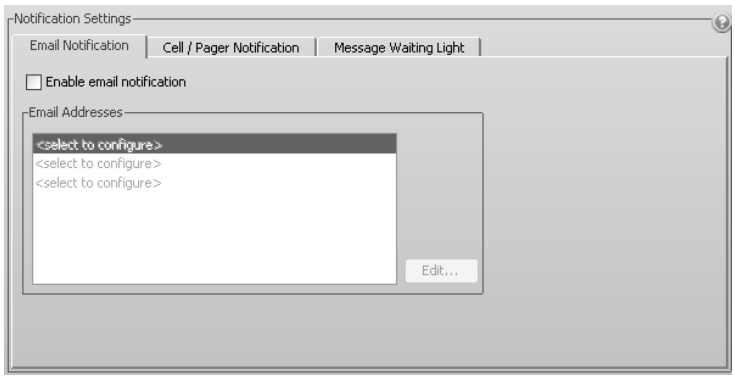
The *Notification Settings* area allows you to set up voicemail notification, which tells the user if a caller leaves a message. The TalkSwitch system can:

- Notify up to four users by e-mail, with the voicemail message included as an attachment.
- Notify a user by phone and/or pager.
- Activate the message waiting light and stutter dial tone on one or more local extensions.
- Perform voicemail screening by routing audio to the PA jack.

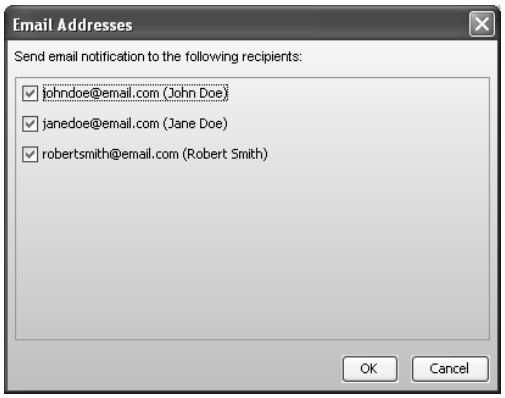
Setting up notification by e-mail

Note that e-mail notifications might be routed to the user's junk or spam folder. Advise each user to set up their e-mail accounts to allow e-mail notifications to go to their inboxes.

1. Set up the e-mail addresses. See *Email Service* on page 40.
2. Select the *Email Notification* tab.



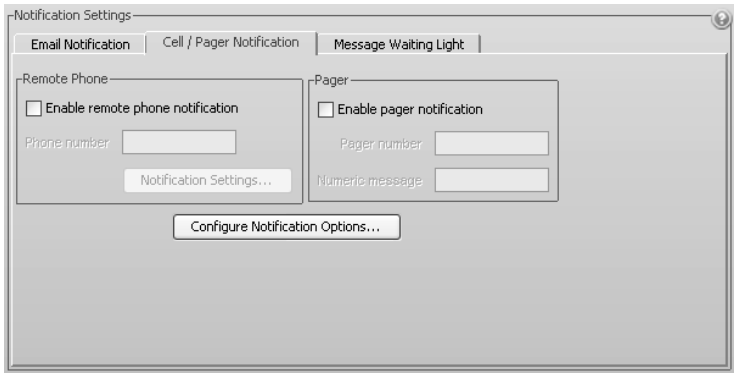
3. Select the *Enable email notification* checkbox.
4. Click **Edit**. The **Email Addresses** window appears.



5. Select up to four e-mail addresses.

Setting up notification by phone

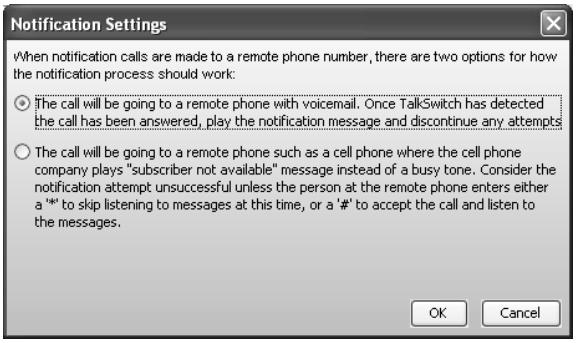
1. Select the *Cell/Pager Notification* tab.



2. Select the *Enable remote phone notification* checkbox.

3. Enter the *Phone number*. Enter the number as you would normally dial it (i.e. without the hunt group number). You can enter digits 0–9, space, dash, comma, **#** and *****. A comma pauses dialing for two seconds.

4. Click **Notification Settings**. The **Notification Settings** window appears.


Notification Settings
When notification calls are made to a remote phone number, there are two options for how the notification process should work:
The call will be going to a remote phone with voicemail. Once TalkSwitch has detected the call has been answered, play the notification message and discontinue any attempts
The call will be going to a remote phone such as a cell phone where the cell phone company plays "subscriber not available" message instead of a busy tone. Consider the notification attempt unsuccessful unless the person at the remote phone enters either a '*' to skip listening to messages at this time, or a '#' to accept the call and listen to the messages.
OK
Cancel

5. Select the notification setting. Choices are:

   - Select the first option to have TalkSwitch play the notification message once, when the phone is answered. TalkSwitch will then consider notification to be successful. Note that if there is any answer other than a busy tone, (e.g. voicemail, *"subscriber not available"* message, etc.) TalkSwitch will consider notification to be successful.

   - Select the second option to have TalkSwitch repeat notification until the user either dials ***** to skip the message, or dials **#** to play the message. TalkSwitch will only consider notification to be successful once the user acknowledges notification by dialing a key. This is useful for cell phones where the telephone company plays a *"subscriber not available"* message instead of a busy tone.

Setting up notification by pager

1. Select the *Cell/Pager Notification* tab.
2. Select the *Enable pager notification* checkbox.
3. Enter the *Pager number*. Enter the number as you would normally dial it (i.e. without the hunt group number). You can enter digits 0–9, space, dash, comma, **#** and *****. A comma pauses dialing for two seconds.
4. Enter the *Numeric message* that will appear on the user's pager.

Setting up the notification options

1. Click **Configure Notification Options**. The **Configure Notification Options** window appears.

2. Select the hunt group the TalkSwitch system will use to phone or page the user.
3. Select the number of times the TalkSwitch system will ring the number before aborting each attempt, ranging from *1 time* to *10 times*.
4. Select the number of additional attempts the TalkSwitch system should make, ranging from *never* to *10 times*.
5. Select the interval between attempts, ranging from *1 minute* to *60 minutes*.
6. Select the modes when the TalkSwitch system will perform notification. Choices include *Mode 1*, *Mode 2*, *Holiday Mode* and *All Modes*.

Setting up message waiting light activation

If a local extension has a new message in its voice mailbox, the TalkSwitch system activates notification on that extension, by default. The user hears a stutter dial tone when they pick up the handset (not applicable to the TS-850i). If the extension supports an FSK message waiting indicator (not applicable to IP phones), the message waiting light flashes. The TS-80, TS-200, TS-400 and TS-600 extensions support a message waiting counter. The display on the extension shows the number of new messages stored in the mailbox(es) associated with that extension.

You can also have the TalkSwitch system activate notification on a local extension if a message is received in any other local extension, remote extension, or general mailbox.

1. Select the *Message Waiting Light* tab.

Notification Settings
Email Notification
Cell / Pager Notification
Message Waiting Light
The TalkSwitch system can turn on the message waiting light at any extension(s) when new messages are left in this mailbox
Extensions to Notify
111
Edit...

2. Click the **Edit** button. The **Message Waiting Indication** window appears.

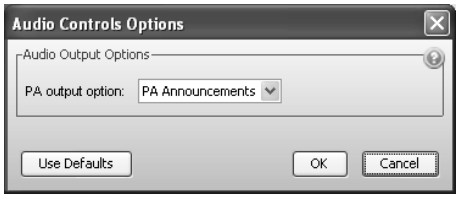
3. Select the extensions.

Voicemail screening

The system can be configured to perform voicemail screening in addition to overhead paging. In this case it will route audio to the PA jack when a caller leaves a voicemail message or a user accesses a voice mailbox. See *PA Output* on page 183.

NOTE: If a user has dialed ***80** to route music on hold through the PA jack, voicemail screening will not interrupt music on hold.

1. Choose **Options > PA Output**. The **Audio Control Options** window appears. The *PA output option* is set to *PA Announcements* by default, which allows overhead paging but not voicemail screening.



2. To enable voicemail screening, select *Voicemail screening*. The system will continue to allow overhead paging, but will also perform voicemail screening.

ADDING IP PHONES

This section describes how to configure an IP phone as a local extension. An IP phone can be internal (located in the office) or external (located outside the office).

We recommend TalkSwitch IP phones and supported third-party IP phones for ease of configuration. TalkSwitch currently supports the following IP phones:

- TalkSwitch 350i and TS-550i. See *Adding TalkSwitch TS-350i and TS-550i IP phones* on page 85 .
- TalkSwitch TS-9112i, TS-9133i and TS-480i. See *Adding TalkSwitch TS-9112i, TS-9133i and TS-480i IP phones* on page 90.
- TalkSwitch TS-850i cordless IP phone. See *Adding TalkSwitch TS-850i IP phones* on page 96.
- Polycom IP 301, IP 320, IP 330, IP 430, IP 501, IP 550, IP 601 and IP 650. See *Adding Polycom IP Phones* on page 102.
- Grandstream GXP2000 and GXP2020. See *Adding Grandstream IP phones* on page 106.
- Counterpath eyeBeam softphone. See *Adding Counterpath IP phones* on page 111.

Visit **http://global.talkswitch.com** for information on which models are available in your region and useful resources such as quick reference cards, user guides and firmware files. To access the IP phone guides, choose **Start > Programs > TalkSwitch 6.12 > Documentation > IP Phone Guides**.

OVERVIEW

For each supported IP phone, this guide describes how to:

- Add the IP phone to the TalkSwitch system configuration.
- Configure the router, if setting up an external IP extension.
- Connect the IP phone to the network.
- Check and update the IP phone firmware version, if required.
- Program the IP phone.

Refer to *Adding Other IP Phones* on page 115 for configuration information of non-supported IP phones.

EXTERNAL IP EXTENSIONS

An external IP extension is an IP phone located outside the office that is configured as a local extension.

External IP extensions can be assigned to any VoIP-equipped TalkSwitch unit that can connect to the phone over a public or private network. See Configuring the router. For additional information see *Connecting to a network* in the *VoIP Network Configuration Guide*.

All VoIP lines are shared by default. If necessary, you can reserve VoIP lines for external IP extensions. See *Reserving VoIP lines* in the *VoIP Network Configuration Guide*.

External IP extensions will use the preferred codec selected for the TalkSwitch profile. The default preferred codec is G.729. You can select a different preferred codec for the TalkSwitch profile. See *Setting code options* in the *VoIP Network Configuration Guide*.

Warning: Calls to an emergency service number using an external IP extension will not send the correct address to the emergency operator. We strongly recommend that you apply a warning label to any external IP extension stating:

If an emergency call is made from this phone, you must provide your address to the emergency operator.

Configuring the router

If you are setting up an external IP extension, you must ensure the router is configured. There should be no adjustments required for the firewall at the remote (phone) location.

The *Router Port Forwarding* area of the **IP Configuration** page displays the type of gateway device (i.e. the type of router), the IP address of the gateway (i.e. router), and whether router configuration is required.

A router is a gateway between the local area network and the Internet. Most routers have a firewall to block unwanted data from the Internet. For voice data to reach the TalkSwitch system through the firewall, port forwarding is required. Port forwarding allows the router to map ports to the IP addresses of the TalkSwitch units. Valid Internet data will use the ports to go through the firewall to the TalkSwitch units.

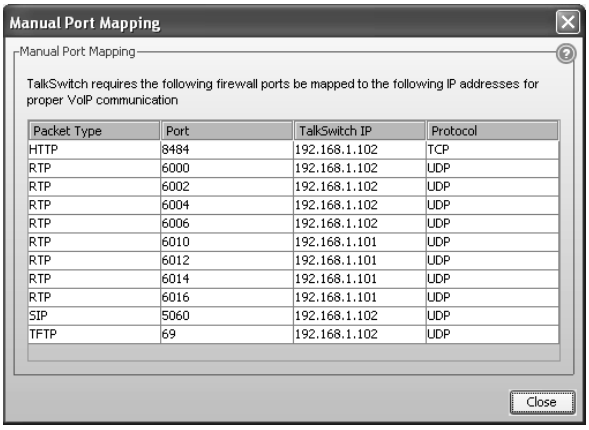
If you are setting up external IP extensions, a TalkSwitch VoIP network, or a VoIP service that doesn't handle port forwarding, port forwarding is required.

If port forwarding is required, and your router supports uPNP (Universal Plug and Play), ensure uPNP is enabled. The TalkSwitch system will use uPNP to automatically set up port forwarding, and the **Automatic (uPNP Enabled)** link will appear. No router configuration is required.

If port forwarding is required but your router doesn't support uPNP, or automatic port forwarding doesn't work, the **Manual Port Mapping Required** link will appear. You will need to configure the router as described below.

1. Select the **IP Configuration** page.

2. If required, click the **Manual Port Mapping Required** link. The **Manual Port Mapping** window appears. It lists the packet type, port number, IP address and protocol of each required port.


Manual Port Mapping
Manual Port Mapping
TalkSwitch requires the following firewall ports be mapped to the following IP addresses for proper VoIP communication
Close

Packet Type	Port	TalkSwitch IP	Protocol
HTTP	8484	192.168.1.102	TCP
RTP	6000	192.168.1.102	UDP
RTP	6002	192.168.1.102	UDP
RTP	6004	192.168.1.102	UDP
RTP	6006	192.168.1.102	UDP
RTP	6010	192.168.1.101	UDP
RTP	6012	192.168.1.101	UDP
RTP	6014	192.168.1.101	UDP
RTP	6016	192.168.1.101	UDP
SIP	5060	192.168.1.102	UDP
TFTP	69	192.168.1.102	UDP

3. To access the router configuration:

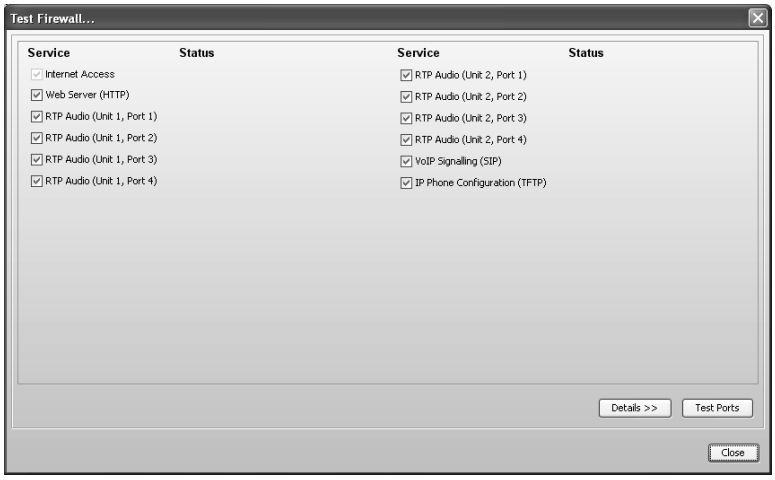
   a) Click the link containing the IP address of the gateway. The default browser starts, and prompts you for the router's user name and password.

   b) Enter the router's user name and password. The browser shows a setup screen.

   c) Navigate to the screen used to set up port forwarding. See your router documentation.

   d) Set up port forwarding using the information from the **Manual Port Mapping** window. See your router documentation for instructions on how to map ports. For information on configuring routers and mapping ports, visit **www.portforward.com/english/routers/port_forwarding/routerindex.htm**.

4. To check the status of each port through the firewall, click **Check Firewall**. The **Test Firewall** window appears.



5. Select the services you want to check.

6. Click **Test Ports**. The system will check the ports for the selected services.

Configuring the router manually

If you cannot access the router configuration through the **IP Configuration** page, configure the router manually.

1. Open the router configuration and navigate to the screen used to set up port forwarding. See your router documentation.
2. Map port 5060 (Type: UDP) to the TalkSwitch unit acting as local proxy. Port 5060 is the default port for forwarding SIP signaling data to the TalkSwitch system.

   If required, you can map a different port. In this case, enter the port in the *SIP signalling port* box of the **VoIP Settings** window.
3. Map the following ports (Type: UDP) to the TalkSwitch units. These RTP ports forward voice traffic to the TalkSwitch system by default.

VoIP Lines/Unit	Unit 1	Unit 2	Unit 3	Unit 4
4	6000–6006	6010–6016	6020–6026	6030–6036
8	6100–6114	6200–6214	6300–6314	6400–6414

   If required, you can map different ports. In this case enter the first port in the *Starting RTP port* box of the **VoIP Settings** window.
4. Map port 69 (Type: UDP) to the TalkSwitch unit acting as local proxy. Port 69 allows the TalkSwitch system to configure external IP extensions.
5. Map port 8484 (Type: TCP) to the TalkSwitch unit acting as local proxy. Port 8484 allows the TalkSwitch system to configure TalkSwitch IP phones as external IP extensions.
6. Map port 9393 (Type: TCP) to the TalkSwitch unit acting as local proxy. Port 9393 allows remote configuration of the TalkSwitch system.
7. If available, enable Quality of Service (QoS) to give voice traffic priority over data.

Save the configuration to the router.

ADDING TALKSWITCH TS-350i AND TS-550i IP PHONES

Connecting the TalkSwitch IP phone to the network

1. Connect a network cable between the LAN port on the phone (marked) and your network (i.e. router or LAN connection). The TalkSwitch 350i and 550i IP phones also have a PC port. The PC port can be used to connect the PC to the network if only one network connection is available.

2. Connect power to the phone, either using the optional power adapter or an 802.3af Power-Over-Ethernet (POE) source.

   If the phone is connected to the same network as the TalkSwitch system, and an extension has been configured with the phone's MAC address, the system will automatically register and configure the phone. When complete, the phone will display the extension name and extension number.

   If the phone is connected to the network and no extension has been configured with the phone's MAC address, the phone will bootup with the model number showing in the display.

Adding the extension to the TalkSwitch system

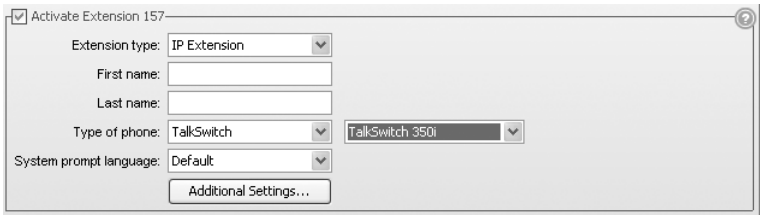
1. Launch the TalkSwitch management software.

2. Select the **Local Extensions/Fax** page.

3. Select an available extension number.

4. Select the *Extension* tab.

Activate extension area



1. Ensure the *Activate extension* check box is selected.

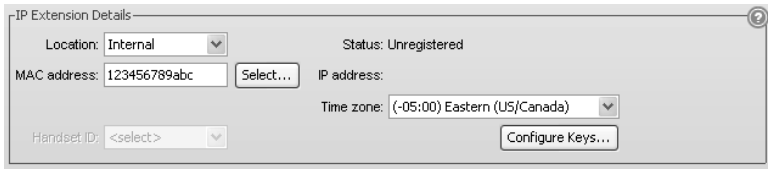
2. If necessary, set the *Extension type* to *IP extension*.

3. Enter the user's *First name* and *Last name*. The names are used for caller ID, and appear within other pages of the TalkSwitch management software. The *Last name* is used with the dial-by-name directory.

4. If necessary, select *TalkSwitch* as the *Type of phone*, and then select the model.

5. Select the language for prompts heard by the user of the extension in the *System prompt language* list. Setting the prompt language will also change the language for text displayed on the phone itself.

IP extension details area

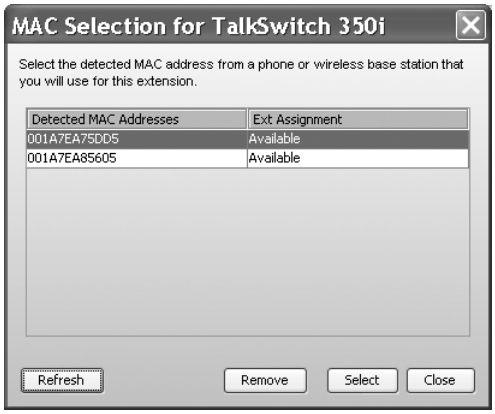


1. Set the *Location*. Choices are:
   - *Internal* — The phone is an internal IP extension located within the office, and is connected to the same LAN as the TalkSwitch unit.
   - *External* — The phone is an external IP extension located outside the office, and is connected to the TalkSwitch unit over the internet. See *External IP extensions* on page 82.

Warning: Calls to an emergency service number using an external IP extension will not send the correct address to the emergency operator. We strongly recommend that you apply a warning label to any external IP extension stating:

If an emergency call is made from this phone, you must provide your address to the emergency operator.

2. If you set *Location* to *External*, select the *Time Zone* that matches the location of the IP phone.
3. Select the *MAC Address* of the phone:
   - You can select the *Phone MAC Address* from a list of automatically-detected phones connected to your LAN. To use this method:
     i) Click the **Select...** button. A **MAC Selection** window appears and lists TalkSwitch IP phones of the selected type.



     ii) Select the MAC address of the IP phone associated to the extension, and then click **Select**.
   - You can enter the *Phone MAC Address* manually. This is the required method if you are setting up an external IP extension. The MAC Address is a 12-digit alphanumeric string located in the barcode on the bottom of the phone and the box the phone came in.

4. Choose **File > Save**. The system will create a configuration file that the phone will download when the phone is restarted.

About programmable function keys

The TS-350i has 6 programmable functions keys, and the TS-550i has 22 programmable keys. The keys allow the user to access TalkSwitch features, and to monitor and engage lines, extensions and queued calls (i.e. line appearance).

Note that the keys cannot be programmed if the extension has hot line access enabled. In this case, the phone will automatically connect the pre-configured resource (external number, extension etc.)

Programming TS-350i function keys

1. Click **Configure Keys**. This button is enabled if the *IP Extension Details* area is complete. The **TalkSwitch 350i** window appears.

2. For each key, select the *Function* and the *Resource* (if applicable). The function keys can be assigned for Line Appearance, Extension Appearance, Queue Appearance, Voicemail, Do Not Disturb (DND), Speed Dial, Park, Unpark, Call Pickup (any or specific extension), Overhead Page, Phone Book configuration, or User Defined (phone). For further details, see *TalkSwitch phone programmable key functions* on page 100.

3. Optionally, you can apply predefined key assignments from a template file using the **Open Template** button. See *Using a key assignment template* on page 88.

   If necessary, use the **Default** button to restore the keys to their default settings.

4. Use the **Print Labels** button to print a label showing the key configuration. Cut out the printed label and insert it in the phone next to the key lights.

5. Your key assignments can be saved as a template for programming additional phones. See *Saving a key assignment template* on page 87.

Saving a key assignment template

You can use the template file as a starting point for the key assignments for another extension. After setting up the key assignments:

1. Click **Save Template As** to save the key assignments to a template file. The **Save Template As** window appears.

2. Enter the filename of the template file, and then click **Save**.

Using a key assignment template

1. To display the key assignments from a template file, click **Open Template**. The **Open Template** window appears.
2. Select the template file, and then click **Open Template**.

Programming TS-550i function keys

1. Click **Configure Keys**. This button is enabled if the *IP Extension Details* area is complete. The **TalkSwitch 550i** window appears.
2. Select up to 22 softkey options. For each option, select the *Function* and the *Resource* (if applicable). The function keys can be assigned for Line Appearance, Extension Appearance, Queue Appearance, Voicemail, Do Not Disturb (DND), Phone Book access, Speed Dial, Park, Unpark, Call Pickup (any or specific extension), Overhead Page, or User Defined (phone). For further details, see *TalkSwitch phone programmable key functions* on page 100.

   If the phone has TS-50e expansion modules installed, use the pull-down menu to select the number of modules installed. Select the tab associated to each module and configure the keys.

3. Optionally, you can apply predefined key assignments from a template file using the **Open Template** button. See *Using a key assignment template* on page 88.

TalkSwitch 550i
TalkSwitch 550i
The 550i has 22 flexible keys that can be assigned to engage TalkSwitch features or monitor and engage lines, extension or queues. The addition of TS-50e Expansion Modules can increase the number of keys to a total of 46.
Number of TS-50e modules attached to this phone:
Base
Module 1
Module 2
Option
Function
Resource
Default
Open Template...
Save Template As...
Print Labels
OK
Cancel

   If necessary, use the **Default** button to restore the keys to their default settings.

4. Use the **Print Labels** button to print a label showing the key configuration. Cut out the printed label and insert it in the phone next to the key lights.
5. Your key assignments can be saved as a template for programming additional phones. See *Saving a key assignment template* on page 87.

Programming the TalkSwitchTS-350i or TS-550i IP phone as an external IP extension

If you are setting up an external IP extension, you must ensure there is a communication path over the network between your phone and your TalkSwitch systsem. See *Configuring the router* on page 82.

Program the following settings into the phone. On the phone:

1. Press the /••• button (TS-350i) / button (TS-550i).
2. Press **3** to enter the **Phone Settings** menu.
3. Press **7** to enter the **Firmware Update** menu.
4. Press **2** to enter the **TFTP Server Address** menu.
5. Use the keypad to enter the public IP address or FQDN of the TalkSwitch system, and then press the **OK** softkey.
6. Press the /••• button (TS-350i) / button (TS-550i) to exit the menu.
7. The phone will request you reboot the phone. Press the **Yes** softkey to initiate the reboot. It should take no more than 1 minute for boot process to finish. During this process, the phone contacts the TalkSwitch unit, which automatically registers and configures the phone. When complete, the phone will display the extension name and extension number. The phone is now ready for use.

If registration does not work

If the IP phone is not successfully registered and configured, perform the following steps. On the phone:

1. Press the /••• button (TS-350i) / button (TS-550i).
2. Press **6** to enter the **default all settings** menu.
3. Press the **Yes** softkey to default the settings.
4. Press the /••• button (TS-350i) / button (TS-550i) to exit the menu.
5. The phone will request you reboot the phone. Press the **Yes** softkey to initiate the reboot. It should take no more than 1 minute for boot process to finish.
6. If setting up an external IP extension, follow the steps in *Programming the TalkSwitch IP phone as an external IP extension* after the phone reboots.
7. Contact your reseller or TalkSwitch Customer Support if registration is still unsuccessful.

ADDING TALKSWITCH TS-9112i, TS-9133i AND TS-480i IP PHONES

To add a TalkSwitch TS-9112i, TS-9133i or TS-480i IP phone within your office, connect the phone to the LAN and follow the prompts on the phone to set the extension number. The phone will work right away. Later you can complete the configuration in the TalkSwitch management software.

Alternatively, you can connect the phone to the network, skip setting the extension number and then do the configuration using the TalkSwitch management software.

Connecting the TS-9112i, TS-9133i and TS-480i IP phones to the network

1. If you will select the extension using the phone itself, ensure the TalkSwitch management software is closed.
2. Connect a network cable between the LAN port on the phone (marked ⧈) and your network (i.e. router or LAN connection). The TalkSwitch TS-9133i and TS-480i IP phones also have a PC port. The PC port can be used to connect the PC to the network if only one network connection is available.
3. Connect power to the phone, either using the power adapter, or an 802.3af Power-Over-Ethernet (POE) source (TS-9133i or TS-480i).

   If the phone is connected to the same network as the TalkSwitch system, and an extension has been configured with the phone's MAC address, the system will automatically register and configure the phone. When complete, the phone will display the extension name and extension number.

   If the phone is connected to a different network than the TalkSwitch system, the phone will display the default time and date of Sat, Jan 1 12:00 am.

   If the phone is connected to the same network as the TalkSwitch system, but no extension has been configured with the phone's MAC address, the following prompt appears on the phone display.

TS-9112i	TS-9133i	TS-480i

`Ext Setup?`
 	
`^=Cancel v=Enter`
	`Ext Setup?`
	
	`^=Cancel v=Enter`
		`Extension Setup?`

4. If you will select the phone's MAC address later, and do not want to select the extension using the phone itself, press the up arrow key on the TS-9112i or TS-9133i, or press the **No** softkey on the TS-480i.
5. To select the extension using the phone itself:

   a) Press the down arrow key on the TS-9112i or TS-9133i, or press the **Yes** softkey on the TS-480i. The phone displays the following prompt.

TS-9112i	TS-9133i	TS-480i

`Ext Number?`
`_`
`^=Cancel v=Enter`
	`Ext Number?`
	`_`
	`^=Cancel v=Enter`
		`Extension Number?`

   b) Use the phone to enter a valid available extension number.

   c) Press the down arrow key on the TS-9112i or TS-9133i, or press the **Enter** softkey on the TS-480i. The phone displays the following prompt.

TS-9112i	TS-9133i	TS-480i
Done! Reboot Now ^=Cancel v=Enter	Done! Reboot Now ^=Cancel v=Enter	Success! Reboot the phone

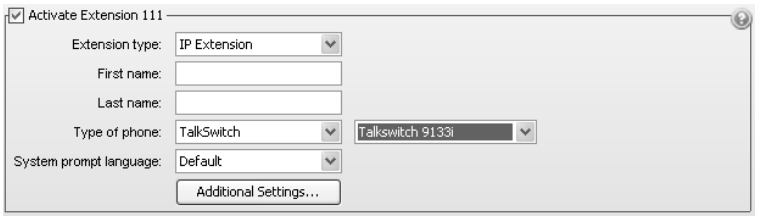
If you have not entered a valid available extension number, the phone returns to step a.

d) Press the down arrow key on the TS-9112i or TS-9133i, or press the **Reboot** softkey on the TS-480i. The phone reboots, and then displays the date, time, extension number, and number of voicemails (if any). The phone is available for use, but you can configure additional features with the TalkSwitch management software.

Adding the extension to the TalkSwitch system

1. Launch the TalkSwitch management software.
2. Select the **Local Extensions/Fax** page.
3. Select an available extension number.
4. Select the *Extension* tab.

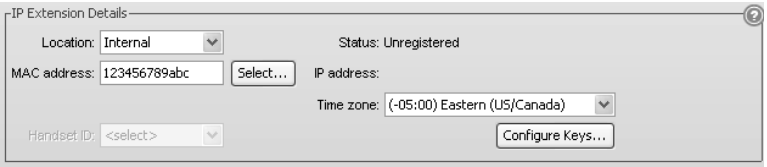
Activate extension area



If you connected a TalkSwitch TS-9112i, TS-9133i or TS-480i IP phone to the LAN and used it to select an extension number, the *Extension type* and *Type of phone* will already be assigned and displayed. In this case you will enter the *First name* and *Last name*, and will select the *System prompt language*.

1. Ensure the *Activate extension* check box is selected.
2. If necessary, set the *Extension type* to *IP extension*.
3. Enter the user's *First name* and *Last name*. The names are used for caller ID, and appear within other pages of the TalkSwitch management software. The *Last name* is used with the dial-by-name directory.
4. If necessary, select *TalkSwitch* as the *Type of phone*, and then select the model.
5. Select the language for prompts heard by the user of the extension in the *System prompt language* list.

IP extension details area



If you connected a TalkSwitch TS-9112i, TS-9133i or TS-480i IP phone to the LAN and used it to select an extension number, the *IP Extension Details* area for that extension will be complete.

If necessary, perform the following steps:

1. Set the *Location*. Choices are:

   - *Internal* — The phone is an internal IP extension located within the office, and is connected to the same LAN as the TalkSwitch unit.
   - *External* — The phone is an external IP extension located outside the office, and is connected to the TalkSwitch unit over the internet. See *External IP extensions* on page 82.

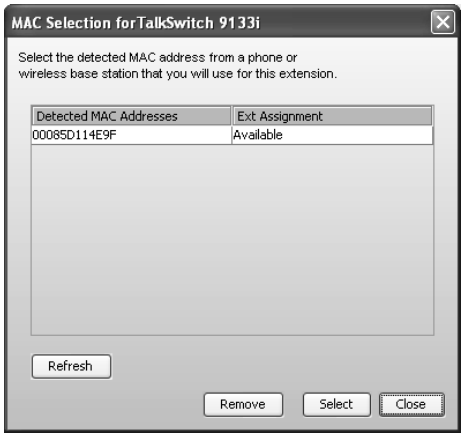
Warning: Calls to an emergency service number using an external IP extension will not send the correct address to the emergency operator. We strongly recommend that you apply a warning label to any external IP extension stating:

If an emergency call is made from this phone, you must provide your address to the emergency operator.

2. If you set *Location* to *External*, select the *Time Zone* that matches the location of the IP phone.
3. Ensure the *Phone MAC Address* is complete:

   - The *Phone MAC Address* will be complete, if you connected the phone to the LAN and used it to select the extension number.
   - You can select the *Phone MAC Address* from a list of automatically-detected phones connected to your LAN. To use this method:

     i) Click the **Select...** button. A **MAC Selection** window appears and lists TalkSwitch IP phones of the selected type.



ii) Select the MAC address of the IP phone for the current extension, and then click **Select**.

- You can enter the *Phone MAC Address* manually. This is the required method if you are setting up an external IP extension. The MAC Address is a 12-digit alphanumeric string located in the barcode on the bottom of the phone and the box the phone came in.

4. Choose **File > Save**. The system will create a configuration file that the phone will download when the phone is restarted.

About programmable function keys

The TS-9133i has 7 programmable functions keys, and the TS-480i has 14 programmable ssoftkeys. The keys allow the user to access TalkSwitch features, and to monitor and engage lines, extensions and queued calls (i.e. line appearance).

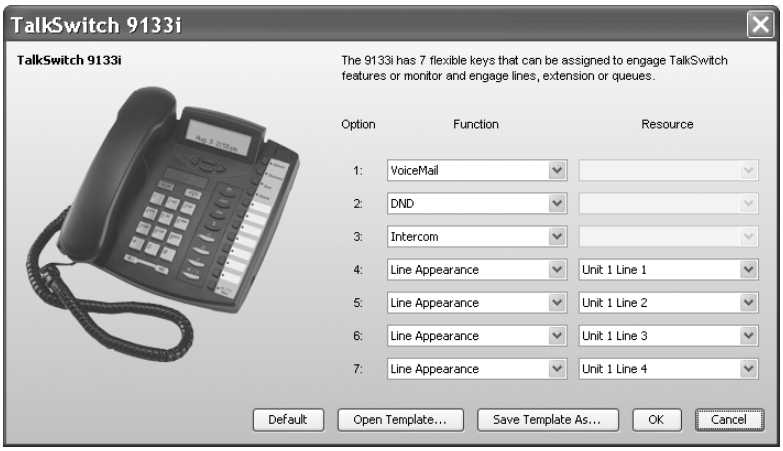
Note that the keys cannot be programmed if the extension has hot line access enabled. In this case, pressing the top function key of the TS-9133i or the **HotLine** softkey of the TS-480i connects to the resource.

Programming TS-9133i function keys

1. Click **Configure Keys**. This button is enabled if the *IP Extension Details* area is complete. The **TalkSwitch 9133i** window appears.

2. For each key, select the *Function* and the *Resource* (if applicable). The function keys can be assigned for Line Appearance, Extension Appearance, Voicemail, Do Not Disturb (DND), Park, Unpark, Call Pickup (any or specific extension), Intercom Page, or Overhead Page. For further details, see *TalkSwitch phone programmable key functions* on page 100.

3. Optionally, you can apply predefined key assignments from a template file using the **Open Template** button. See *Using a key assignment template* on page 88.



If necessary, use the **Default** button to restore the keys to their default settings.

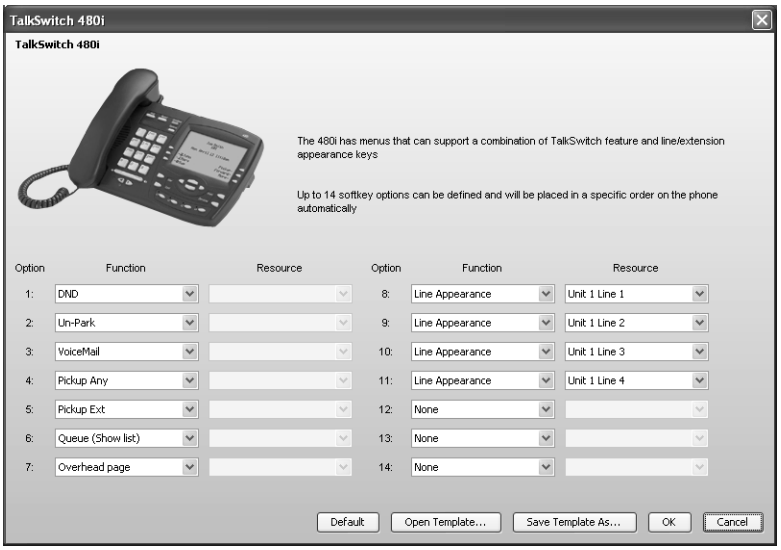
4. Your key assignments can be saved as a template for programming additional phones. See *Saving a key assignment template* on page 87.

Programming TS-480i softkeys

1. Click **Configure Keys**. This button is enabled if the *IP Extension Details* area is complete. The **TalkSwitch 480i** window appears.

2. Select up to 14 softkey options. For each option, select the *Function* and the *Resource* (if applicable). The function keys can be assigned for Line Appearance, Extension Appearance, Queue Appearance, Voicemail, Do Not Disturb (DND), Park, Unpark, Call Pickup (any or specific extension), show Queue, or Overhead Page. For further details, see *TalkSwitch phone programmable key functions* on page 100.

3. Optionally, you can apply predefined key assignments from a template file using the **Open Template** button. See *Using a key assignment template* on page 88.


TalkSwitch 480i
TalkSwitch 480i
The 480i has menus that can support a combination of TalkSwitch feature and line/extension appearance keys
Up to 14 softkey options can be defined and will be placed in a specific order on the phone automatically
Option Function Resource Option Function Resource
1: DND
2: Un-Park
3: VoiceMail
4: Pickup Any
5: Pickup Ext
6: Queue (Show list)
7: Overhead page
8: Line Appearance Unit 1 Line 1
9: Line Appearance Unit 1 Line 2
10: Line Appearance Unit 1 Line 3
11: Line Appearance Unit 1 Line 4
12: None
13: None
14: None
Default
Open Template...
Save Template As...
OK
Cancel

   If necessary, use the **Default** button to restore the keys to their default settings.

4. Your key assignments can be saved as a template for programming additional phones. See *Saving a key assignment template* on page 87.

Programming the TalkSwitch IP phone as an external IP extension

If you are setting up an external IP extension, you must ensure there is a communication path over the network between your phone and your TalkSwitch system. See *Configuring the router* on page 82.

Program the following settings into the phone. On the phone:

1. Press the **Options** key.

2. Use the cursor keys to navigate to the **Network Settings** menu, and then press **Enter** or **Show**.

3. Enter the password. By default this is 22222. When the TalkSwitch unit configures the phone, it changes the password to 23646 (which spells “admin” on the telephone keypad).

4. Use the cursor keys to navigate to the **TFTP Settings** menu, and then press **Enter** or **Show**.

5. Select **Primary TFTP**, and then press **Enter** or **Show**.

6. Use the keypad to enter the public IP address or FQDN of the TalkSwitch system, and then press **Set** or **Done**.

7. Reboot the phone. After a short delay, the phone will indicate it is initializing the network, checking for firmware updates, and following automatic programming steps. During this process the phone contacts the TalkSwitch unit, which automatically registers and configures the phone. When complete, the phone will display the extension name and extension number. The phone is now ready for use.

If registration does not work

If the IP phone is not successfully registered and configured, perform the following steps. On the phone:

1. Press the **Options** key.
2. Use the cursor keys to navigate to the **Phone Status** menu, and then press **Enter** or **Show**.
3. Use the cursor keys to navigate to the **Restore Defaults** menu, and then press **Enter** or **Show**.
4. Enter the password. By default this is 22222. When the TalkSwitch unit configures the phone, it changes the password to 23646 (which spells "admin" on the telephone keypad).
5. Use the cursor keys to navigate to the **Factory Defaults** menu, and then press **Enter** or **Show**.
6. Select **All Defaults**, and then press **Enter** or **Show**.
7. If setting up an external IP extension, follow the steps in *Programming the TalkSwitch IP phone as an external IP extension* on page 94 after the phone reboots.
8. Contact your reseller or TalkSwitch Customer Support if registration is still unsuccessful.

Further configuration

The steps above will configure your TalkSwitch IP phone and will enable the programmable keys.

Click the *Local Extension/Fax* Help icons for instructions on configuring direct line access, hunt group permissions, Caller ID settings (VoIP only), call handling, and voicemail.

For phone-specific configuration, refer to your phone user guide.

Contact your reseller or TalkSwitch Customer Support if you require further assistance with special ports or network settings.

ADDING TALKSWITCH TS-850i IP PHONES

Installing the base

1. Install the TS-850i base according to the instructions in the *TS-850i Start Guide*.
2. Connect the Ethernet cable from the WAN port on the base to a LAN port on your router or Ethernet switch. The TalkSwitch unit must be connected to the same LAN.
3. Install the TS-850i handset(s) and allow them to register with the TS-850i, according to the instructions in the *TS-850i Start Guide*.

Configuring the TS-850i as an IP extension

After installing the TS-850i, configure your phone as an IP extension. The phone can be located within the office, or outside the office (i.e. as an external IP extension).

Before configuring a phone, ensure that no handsets registered with the associated base are being used. If a handset is on a call, the call will be terminated when the system resets the base during configuration of the phone.

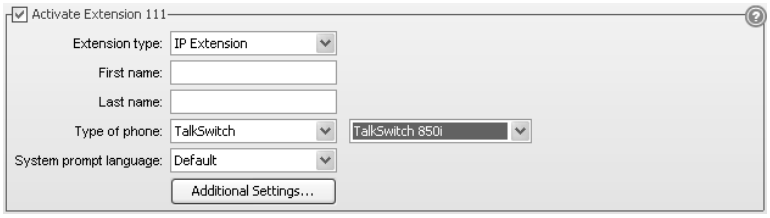
NOTE: The TS-850i requires version 6.11 or greater of the TalkSwitch management software.

Configuring the TS-850i for use within the office

If the TS-850i is located within the office, it connects to your TalkSwitch system over the local area network (LAN).

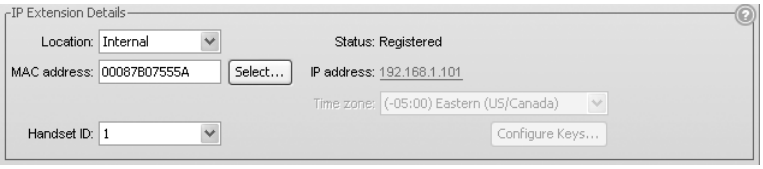
1. Launch the TalkSwitch management software.
2. Select the **Local Extensions/Fax** page.
3. Select an available extension number.
4. Select the *Extension* tab.

Activate extension area



1. Ensure the *Activate extension* check box is selected.
2. Set the *Extension type* to *IP extension*.
3. Enter the user's *First name* and *Last name*. The names are used for caller ID, and appear within other pages of the TalkSwitch management software. The *Last name* is used with the dial-by-name directory.
4. Select *TalkSwitch* as the *Type of phone*, and then select *TalkSwitch 850i* as the model.
5. Select the language for prompts heard by the user of the extension in the *System prompt language* list.

IP extension details area



1. Set the *Location*. Choices are:
   - *Internal* — The phone is an internal IP extension located within the office, and is connected to the same LAN as the TalkSwitch unit.
   - *External* — The phone is an external IP extension located outside the office, and is connected to the TalkSwitch unit over the internet. See *External IP extensions* on page 82.

Warning: Calls to an emergency service number using an external IP extension will not send the correct address to the emergency operator. We strongly recommend that you apply a warning label to any external IP extension stating:

If an emergency call is made from this phone, you must provide your address to the emergency operator.

2. If you set *Location* to *External*, select the *Time Zone* that matches the location of the IP phone.
3. Select the *MAC Address* of the phone:
   - You can select the *Phone MAC Address* from a list of automatically-detected phones connected to your LAN. To use this method:
     i) Click the **Select...** button. A **MAC Selection** window appears and lists TalkSwitch IP phones of the selected type.

     ii) Select the MAC address of the IP phone associated to the extension, and then click **Select**.
   - You can enter the *Phone MAC Address* manually. This is the required method if you are setting up an external IP extension. The MAC Address is a 12-digit alphanumeric string located in the barcode on the bottom of the phone and the box the phone came in.

   Note that multiple IP extensions can share the same base.

4. Select the *Handset ID* for the extension. Use the handset name as shown on the handset screen.

5. After configuring the extensions, choose **File > Save**. The TalkSwitch system will automatically configure each handset.

Programming the TalkSwitch TS-850i IP phone as an external IP extension

If you are setting up an external IP extension, you must ensure there is a communication path over the network between your phone and your TalkSwitch systsem. See *Configuring the router* on page 82.

Before configuring a phone, ensure that no handsets registered with the associated base are being used. If a handset is on a call, the call will be terminated when the system resets the base during configuration of the phone.

1. In Idle mode, press the **OK** key on the handset to access the Main menu.

2. Choose **Settings**, and then press the **OK** key.

3. Choose **Network settings**, and then press the **OK** key.

4. Select **Provision server**, and then press the **OK** key.

5. Enter the Public IP address or FQDN (fully qualified domain name) of the TalkSwitch system as provided by your administrator. To enter a period:

   a) Press the **Options** softkey.

   b) Choose **Insert symbol**, and then press the **OK** key.

   c) Use the navigation keys to choose the period, and then press the **OK** key.

6. Press the **Save** softkey to accept the changes, and then press the **Back** key.

7. When prompted to reboot the base, select **Yes**. The TS-850i will automatically download the configuration from the TalkSwitch system.

Using the TS-850i with a traditional phone line

NOTE: If you connected the telephone cable from the Line port on the base to the telephone wall socket, you can use the TS-850i with a traditional phone line.

When connected to the traditional telephone network (PSTN) or to a TalkSwitch system as an analog extension, FLASH is engaged by pressing the **R#** key. The key MUST be held for approximately 2 seconds until you connect to the other caller (call waiting) or receive dial tone to initiate a calling feature.

Further configuration

The steps above will configure your TalkSwitch IP phone and will enable the programmable keys. Click the *Local Extension/Fax* Help icons for instructions on configuring direct line access, hunt group permissions, Caller ID settings (VoIP only), call handling, and voicemail.

For phone-specific configuration, refer to your phone user guide.

Contact your reseller or TalkSwitch Customer Support if you require further assistance with special ports or network settings.

Checking and updating firmware

Occasionally, you may want to check and/or update the TS-850i firmware.

To check the firmware version:

1. In Idle mode, press the **Increase Volume** button on the left side of the handset. After a brief delay the phone will display a System Status screen.

2. Press the down arrow button to show the second screen, which includes the firmware version (next to the title "SW:").

NOTE: If your installation uses repeaters to extend the range of the handsets, turn the repeaters off and bring the handsets within range of the bases before updating the firmware.

For instructions on obtaining firmware for your region and language, contact your Authorized TalkSwitch reseller. If you purchased directly from TalkSwitch, visit **http://global.talkswitch.com** to obtain further information or assistance.

For more information, refer to the Troubleshooting section in your phone user guide.

TALKSWITCH PHONE PROGRAMMABLE KEY FUNCTIONS

Many TalkSwitch IP phone models offer programmable keys. The function and associated resources are assigned using TalkSwitch Management Software in the **Local Extensions/Fax > IP Extension Details > Configure Keys** page. Supported functions for a phone model typically include most of the items listed below.

- *Line appearance* — Select a telephone line or VoIP number as the resource. The corresponding button or softkey will display the status of the line, and allow you to make calls with a single press of the button or softkey.

  For phones equipped with button lights, the button will light up when the line is in use, flash if the line is ringing, or be off when the line is available.

  For phones with softkeys, a status icon and line ID will be displayed beside the softkey. The display will show an off-hook icon when the line is in use, show a ringing icon when the line is ringing, and show an on-hook icon when the line is available.

- *Extension appearance* — Select a local extension as the resource.

  The corresponding button or softkey will display the status of the selected extension, and allow you to call the extension with a single press of the button or softkey.

  For phones equipped with button lights, the button will light up when the selected extension is in use, flash if the extension is ringing, or be off when the extension is available.

  For phones with softkeys, a status icon and extension call ID will be displayed beside the softkey. The display will show an off-hook icon when the selected extension is in use, show a ringing icon when the extension is ringing, and show an on-hook icon when the extension is available. The extension call ID is also be displayed.

- *Queue appearance* — Select a local extension as the resource. The corresponding button or softkey will indicate whether calls are queued at the selected extension, and allow you to pick up the oldest queued call with a single press of the button or softkey.

  For phones equipped with button lights, the button will flash if calls are queued for the extension, or be off when there are no queued calls.

  For phones with softkeys, a status icon, extension number and "Q" will be displayed beside the softkey. The display will show a ringing icon if calls are queued for the extension, or an on-hook icon when there are no queued calls.

- *Voicemail* — Do not select a resource. Press the button or softkey to access the voice mailbox of the local extension. Lights or icons are not used for voicemail keys. Note: You can also access Voicemail by pressing ****#**.
- *DND* — Do not select a resource. Press the button or softkey to toggle Do Not Disturb mode on or off. Lights or icons are not used for DND keys. Note: You can also toggle DND mode by pressing ***62#**.
- *Park* — Do not select a resource. Press the button or softkey to put the call on hold, in the next available park orbit. The system will respond with the park orbit number (500 to 509). Lights or icons are not used for Park keys. Note: You can also Park a call by pressing /***510#** on a TS-350i, or ***510#** on a TS-550i, or Xfer ***510#** on a TS-9112i, TS-9133i or TS-480i.
- *Un-park* — Do not select a resource. Press the button or softkey, select the park orbit number (500 to 509), then press Unpark to retrieve the call. Softkey displays will show the parked call ID beside the orbit number. Lights or icons are not used for Unpark keys. Note: You can also Unpark a call by lifting the handset, pressing ******, dialing the park orbit number, then pressing **#**.

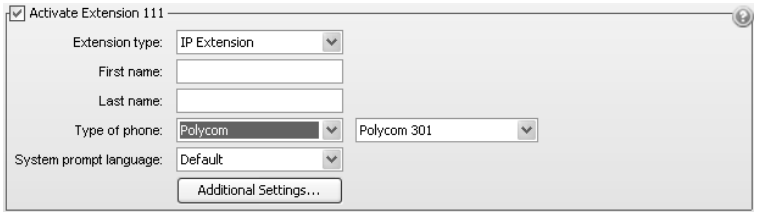
- *Pickup any* — Do not select a resource. Press the button or softkey to answer a call from an outside number ringing any extension. Lights or icons are not used for Pickup keys. Note: You can also pick up a call by pressing ***9#**.
- *Pickup ext* — Do not select a resource. Press the button or softkey, dial an extension, then dial **#** (or softkey again) to answer a call from an outside number or extension ringing the selected extension. Lights or icons are not used for Pickup keys. Note: You can also pick up a call at a selected extension by pressing ***7**, dialing the extension, then pressing **#**.
- *Intercom page* — Do not select a resource. Press the button or softkey, dial an extension, then press **#** to page the extension in Intercom mode. The intercom page function can only page TalkSwitch phones. Lights or icons are not used for Page keys. Note: You can also page a TalkSwitch phone by pressing ***84**, dialing the extension, then pressing **#**, or the **Dial** key or softkey.
- *Overhead page* — Do not select a resource. Press the button or softkey to connect to an attached external PA system. Lights or icons are not used for Page keys. Note: You can also page the external PA system by pressing ***0#**.
- *Queue (show list)* — Do not select a resource. Press the **Queue** softkey, select a queued call, then press **Retrieve** to connect to the selected call. This function is only available when programmed as a function key on supported phone models (e.g. TS-480i).
- *Phone book access* — Select the phone book record two-digit speed dial number as the resource. Press the button or softkey to place a call using the contact information from the associated phone book record. When the button is pressed, it will light up for the duration of the call. This function is available on the TS-350i and TS-550i models.
- *User Defined (phone)* — The button or softkey function is assigned using the phone's configuration options, and is not assigned using TalkSwitch Management software.
- *None* — The button or softkey is not programmed.

ADDING POLYCOM IP PHONES

Adding the extension to the TalkSwitch system

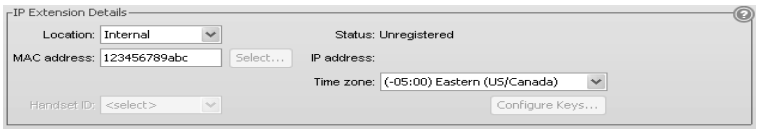
1. Launch the TalkSwitch management software.
2. Select the **Local Extension/Fax** page.
3. Select the *Extension* tab.
4. Select an available extension number.

Activate extension area



1. Select the *Activate Extension* checkbox.
2. Set the *Extension type* to *IP extension*.
3. Enter the user's *First name* and *Last name*. The names are used for caller ID, and appear within other pages of the TalkSwitch management software. The *Last name* is used with the dial-by-name directory.
4. Select *Polycom* as the *Type of phone*, and then select the model. Choices are *Polycom 301*, *Polycom 320*, *Polycom 430*, *Polycom 501*, *Polycom 550*, *Polycom 601* and *Polycom 650*.
5. Select the language for prompts heard by the user of the extension in the *System prompt language* list.
6. Click the **Additional Settings** button to set up:
   - Direct line access (see *Direct line access* on page 61)
   - Hunt group permissions (see *Hunt group permissions* on page 64)
   - Caller ID settings (see *Caller ID settings* on page 65)

IP extension details area



1. Set the *Location*. Choices are:
   - *Internal* — The phone is an internal IP extension located within the office, and is connected to the same LAN as the TalkSwitch unit.

- *External* — The phone is an external IP extension located outside the office, and is connected to the TalkSwitch unit over the internet. A VoIP-enabled TalkSwitch unit is required.

Warning: Using an external IP extension to call an emergency service number will not send the correct address to the emergency operator. We strongly recommend that you apply a warning label to any external IP extension stating:

If an emergency call is made from this phone, you must provide your address to the emergency operator.

2. If you are setting up an external IP extension, select the *Time Zone* of the IP phone.
3. Enter the *Phone MAC Address*. This is a 12-digit alphanumeric string located in the barcode. The MAC address allows the TalkSwitch unit to connect to the IP phone and automatically configure it.

4. See *Setting up call handling on page 67* and *Voicemail tab on page 75* for instructions on configuring call handling and voicemail.
5. Choose **File > Save**. The system will create a configuration file that the phone will download when the phone is restarted.

Configuring the system for external IP extensions

If you are setting up an external IP extension:

1. Ensure your TalkSwitch system is connected to a network. See *Connecting to a network on page 231*.
2. Ensure that you have set up a public IP address for the TalkSwitch system. See *Public IP address* on page 35.
3. Ensure the router is configured at the TalkSwitch location. There should be no need to make any adjustments to the firewall at the remote location. See *Router port forwarding* on page 36.
4. All VoIP lines are shared by default. You can also reserve VoIP lines for external IP extensions. See *Line reservation* on page 50.
5. External IP extensions will use the preferred codec selected for the TalkSwitch profile. The default preferred codec is G.729. You can select a different preferred codec for the TalkSwitch profile. See *Setting codec options* on page 55.

Connecting the Polycom phone to the network

1. Connect a network cable between your router and the port marked LAN on the back of the phone. The PC port on the phone can be used to connect the PC if only one LAN port is available.
2. Connect the power adapter to the phone.

Confirming the Polycom firmware version

The TalkSwitch system requires the IP phone to have a specific firmware version.

1. Check the required firmware version in the support section at **http://global.talkswitch.com**.
2. Confirm the firmware version currently installed the IP phone. On the phone:
   a) Press the **Menu** key and select **Status > Platform > Application > Main**.
   b) Compare *Version* to the required firmware version. If the firmware version is the same, proceed to *Programming the Polycom IP phone* on page 104. If not, follow the instructions in *Updating the Polycom firmware* on page 104.

Programming the Polycom IP phone

The TalkSwitch system automatically configures the necessary parameters required for the proper operation of the Polycom phone with TalkSwitch. On the phone:

1. Press the **Menu** key and select **Settings > Advanced**.
2. Enter the password, and then press the **Enter** key. By default this is 456. When the TalkSwitch unit configures the phone, it changes the password to 23646 (which spells "admin" on the telephone keypad).
3. Select **Admin Settings > Network Configuration**.
4. Select **Server Menu** and change *ServerType* to *TrivialFTP*.
5. Select **Server Address** and press the **Edit** key. Press the **1/A/a** key. If you are setting up an internal IP extension, enter the IP address of the TalkSwitch unit acting as local proxy. If you are setting up an external IP extension, enter the public IP address or FQDN of the TalkSwitch system. Use the * key to enter decimal points.
6. Press the **Exit** button three times. **Advanced** appears at the top of the menu.
7. Select **Restart Phone** and confirm by selecting **Yes**.

The phone connects to the TalkSwitch local proxy and downloads the configuration file. If the configuration is successful, the phone will reboot and show the time, date and extension number. To ensure full functionality, dial another extension and check the audio transmission in both directions.

Updating the Polycom firmware

This procedure is only required if you need to update the Polycom firmware. To update the firmware:

- Install a TFTP application.
- Download and extract firmware files.
- Configure the phone to point to the firmware files.

Installing a TFTP application

You require a TFTP server application to update the phone firmware. One such free TFTP utility can be downloaded from Solarwinds at the following URL:

http://support.solarwinds.net/updates/New-customerFree.cfm?ProdId=52

1. Download and install the TFTP program.
2. Take note of the location of the TFTP program root directory location. The phone firmware files are extracted to this directory in the next section. (i.e. for Solarwinds TFTP, the root directory is **C:\TFTP-Root**).

Downloading and extracting firmware files

Firmware files for supported IP phones can be found in the support section at **http://global.talkswitch.com**.

1. Download the firmware file required for your phone from the URL above. The files are contained in ZIP archives.
2. Extract the firmware files into the TFTP program root directory location.

Configuring the phone to point to the firmware files

Changing the phone **Server Address** setting to the IP address of the PC with the TFTP program and the firmware files enables the update process. On the phone:

1. Press the **Menu** key and select **Settings > Advanced**.
2. Enter the password, and then press the **Enter** key. By default this is 456. When the TalkSwitch unit configures the phone, it changes the password to 23646 (which spells “admin” on the telephone keypad).
3. Select **Admin Settings > Network Settings**.
4. Select **Server Menu** and change *ServerType* to *TrivialFTP*.
5. Select **Server Address** and press the **Edit** key. Press the **1/A/a** key and enter the IP address of the PC running the TFTP server, using the * key to enter decimal points.
6. Press the **Exit** button three times. **Advanced** appears at the top of the menu.
7. Press **3** for **Restart Phone** and confirm by pressing the **Yes** button.
8. Once the firmware is updated, reconfigure the phone according to *Programming the Polycom IP phone* on page 104. Note that some errors will occur for files not found. These are expected and can be ignored.

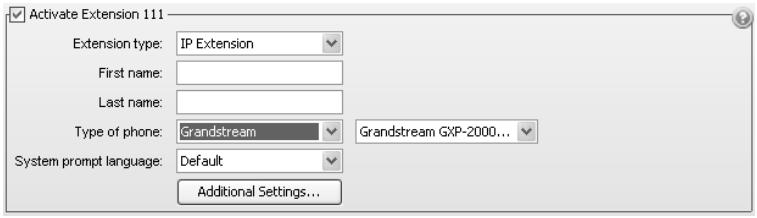
For more information, refer to the Troubleshooting section in your phone user guide.

ADDING GRANDSTREAM IP PHONES

Adding the extension to the TalkSwitch system

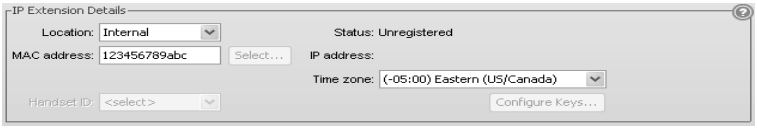
1. Launch the TalkSwitch management software.
2. Select the **Local Extension/Fax** page.
3. Select the *Extension* tab.
4. Select an available extension number.

Activate extension area



1. Select the *Activate Extension* checkbox.
2. Set the *Extension type* to *IP extension*.
3. Enter the user's *First name* and *Last name*. The names are used for caller ID, and appear within other pages of the TalkSwitch management software. The *Last name* is used with the dial-by-name directory.
4. Select *Grandstream* as the *Type of phone*, and then select *Grandstream GXP-2000/2020*.
5. Select the language for prompts heard by the user of the extension in the *System prompt language* list.
6. Click the **Additional Settings** button to set up:
   - Direct line access (see *Direct line access* on page 61)
   - Hunt group permissions (see *Hunt group permissions* on page 64)
   - Caller ID settings (see *Caller ID settings* on page 65)

IP extension details area



1. Set the *Location*. Choices are:
   - *Internal* — The phone is an internal IP extension located within the office, and is connected to the same LAN as the TalkSwitch unit.

- *External* — The phone is an external IP extension located outside the office, and is connected to the TalkSwitch unit over the internet. A VoIP-enabled TalkSwitch unit is required.

Warning: Using an external IP extension to call an emergency service number will not send the correct address to the emergency operator. We strongly recommend that you apply a warning label to any external IP extension stating:

If an emergency call is made from this phone, you must provide your address to the emergency operator.

2. If you are setting up an external IP extension, select the *Time Zone* of the IP phone.
3. Enter the *Phone MAC Address*. This is a 12-digit alphanumeric string located below the second barcode. The barcode label is on the bottom of the phone. The MAC address allows the TalkSwitch unit to connect to the IP phone and then automatically configure it.

4. See *Setting up call handling on page 67* and *Voicemail tab on page 75* for instructions on configuring call handling and voicemail.
5. Choose **File > Save**. The system will create a configuration file that the phone will download when the phone is restarted.

Configuring the system for external IP extensions

If you are setting up an external IP extension:

1. Ensure your TalkSwitch system is connected to a network. See *Connecting to a network on page 231*.
2. Ensure that you have set up a public IP address for the TalkSwitch system. See *Public IP address* on page 35.
3. Ensure the router is configured at the TalkSwitch location. There should be no need to make any adjustments to the firewall at the remote location. See *Router port forwarding* on page 36.
4. All VoIP lines are shared by default. You can also reserve VoIP lines for external IP extensions. See *Line reservation* on page 50.
5. External IP extensions will use the preferred codec selected for the TalkSwitch profile. The default preferred codec is G.729. You can select a different preferred codec for the TalkSwitch profile. See *Setting codec options* on page 55.

Connecting the Grandstream phone to the network

1. Connect a network cable between your router and the port marked LAN on the back of the phone. The PC port on the phone can be used to connect the PC if only one LAN port is available.
2. Connect the power adapter to the phone.

3. Once the phone boots up, it will attempt to obtain an IP address from a router DHCP server.

Confirming the Grandstream firmware version

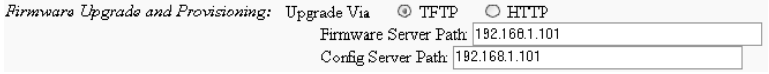
The TalkSwitch system requires the IP phone to have a certain firmware version.

1. Check the required firmware version in the support section at **http://global.talkswitch.com**.
2. Confirm the firmware version currently installed the IP phone. On the phone:
   a) Press the round key (surrounded by four arrow keys) to access the menu.
   b) Select **Status** and press the round key again.
   c) Scroll down until you see **Prog**.
   d) Compare **Prog** to the required firmware version. If the firmware version is the same, proceed to *Programming a Grandstream IP phone* on page 108. If not, follow the instructions in *Updating the Grandstream firmware* on page 109.

Programming a Grandstream IP phone

1. Note the IP address that appears on the screen of the Grandstream phone.
2. In a web browser, enter the IP address in the *Address* field.
3. Enter the password. By default this is *admin*. When the TalkSwitch unit configures the phone, it changes the password to 23646 (which spells "admin" on the telephone keypad). The **Grandstream Device Configuration** page appears.

4. On the **Grandstream Device Configuration** page, click **Advanced**.



5. In the section for *Firmware Upgrade and Provisioning*, set *Upgrade Via* to *TFTP*.
6. If you are setting up an internal IP extension, set *Config Server Path* to the IP address of the TalkSwitch unit acting as local proxy. If you are setting up an external IP extension, set *Config Server Path* to the public IP address or FQDN of the TalkSwitch system.
7. Scroll to the bottom of the page and click the **Update** button to save the settings.

8. On the following screen, click the **Reboot** button to apply the settings.

Note: The reboot process will be completed when you see the extension number and IP address on the display of the phone. When the phone successfully registers, you will see the filled-in Ethernet icon.

9. Test to ensure the phone is configured properly by dialing another local extension.

Updating the Grandstream firmware

This procedure is only required if you need to update the Grandstream firmware. To update the firmware:

- Install a TFTP application.
- Download and extract firmware files.
- Configure the phone to point to the firmware files.

Installing a TFTP application

You require a TFTP server application to update the phone firmware. One such free TFTP utility can be downloaded from Solarwinds at the following URL:

http://support.solarwinds.net/updates/New-customerFree.cfm?ProdId=52

1. Download and install the TFTP program.
2. Take note of the location of the TFTP program root directory location. The phone firmware files are extracted to this directory in the next section. (i.e. for Solarwinds TFTP, the root directory is **C:\TFTP-Root**).

Downloading and extracting firmware files

Firmware files for supported IP phones can be found in the support section at **http://global.talkswitch.com**.

1. Download the firmware file required for your phone from the URL above. The files are contained in ZIP archives.
2. Extract the firmware files into the TFTP program root directory location.

Configuring the phone to point to the firmware files

Changing the phone *Firmware Server Path* setting to the IP address of the PC with the TFTP program and the firmware files enables the update process.

1. Note the IP address that appears on the screen of the Grandstream phone.
2. In a web browser, enter the IP address in the *Address* field.
3. Enter the password. By default this is *admin*. When the TalkSwitch unit configures the phone, it changes the password to 23646 (which spells "admin" on the telephone keypad). The **Grandstream Device Configuration** page appears.

4. On the **Grandstream Device Configuration** page, click **Advanced**.

5. In the section for *Firmware Upgrade and Provisioning*, set *Upgrade Via* to *TFTP*.

6. Set *Firmware Server Path* to the IP address of the PC running the TFTP server.
7. Scroll to the bottom of the page and click the **Update** button to save the settings.

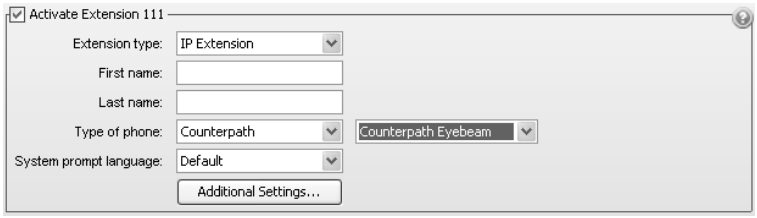
8. On the following screen, click the **Reboot** button to apply the settings.

ADDING COUNTERPATH IP PHONES

Adding the extension to the TalkSwitch system

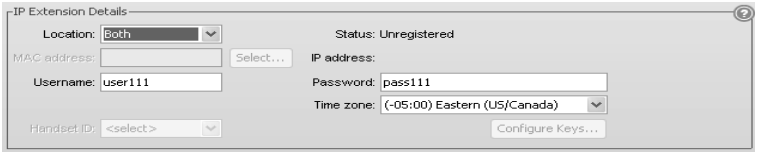
1. Launch the TalkSwitch management software.
2. Select the **Local Extension/Fax** page.
3. Select the *Extension* tab.
4. Select an available extension number.

Activate extension area



1. Select the *Activate Extension* checkbox.
2. Set the *Extension type* to *IP extension*.
3. Enter the user's *First name* and *Last name*. The names are used for caller ID, and appear within other pages of the TalkSwitch management software. The *Last name* is used with the dial-by-name directory.
4. Select *Counterpath* as the *Type of phone*, and then select *Counterpath Eyebeam*.
5. Select the language for prompts heard by the user of the extension in the *System prompt language* list.
6. Click the **Additional Settings** button to set up:
   - Direct line access (see *Direct line access* on page 61)
   - Hunt group permissions (see *Hunt group permissions* on page 64)
   - Caller ID settings (see *Caller ID settings* on page 65)

IP extension details area



1. Set the *Location*. Choices are:
   - *Internal* — The phone is an internal IP extension located within the office, and is connected to the same LAN as the TalkSwitch unit.
   - *External* — The phone is an external IP extension located outside the office, and is connected to the TalkSwitch unit over the internet. A VoIP-enabled TalkSwitch unit is required.

- *Both* — The phone can be used as an internal or external IP extension. A VoIP-enabled TalkSwitch unit is required.

Warning: Using an external IP extension to call an emergency service number will not send the correct address to the emergency operator. We strongly recommend that you apply a warning label to any external IP extension stating:

If an emergency call is made from this phone, you must provide your address to the emergency operator.

2. If the phone can be used as an external IP extension, select the *Time Zone* of the IP phone.
3. Enter the *Username* and *Password*. The default settings are user[extension number] and pass[extension number].
4. See *Setting up call handling on page 67* and *Voicemail tab on page 75* for instructions on configuring call handling and voicemail.
5. Choose **File > Save**. The system will create a configuration file that the phone will download when the phone is restarted.

Configuring the system for external IP extensions

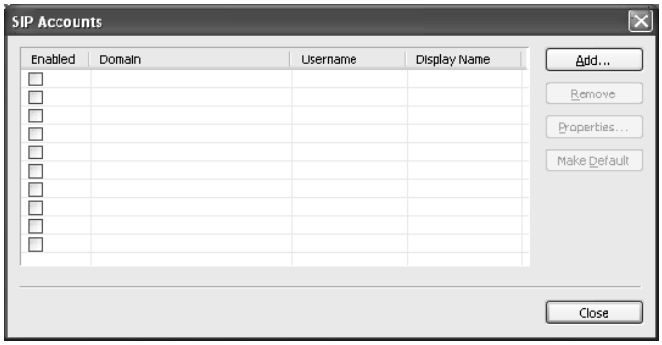
If you are setting up an external IP extension:

1. Ensure your TalkSwitch system is connected to a network. See *Connecting to a network on page 231*.
2. Ensure that you have set up a public IP address for the TalkSwitch system. See *Public IP address* on page 35.
3. Ensure the router is configured at the TalkSwitch location. There should be no need to make any adjustments to the firewall at the remote location. See *Router port forwarding* on page 36.
4. All VoIP lines are shared by default. You can also reserve VoIP lines for external IP extensions. See *Line reservation* on page 50.
5. External IP extensions will use the preferred codec selected for the TalkSwitch profile. The default preferred codec is G.729. You can select a different preferred codec for the TalkSwitch profile. See *Setting codec options* on page 55.

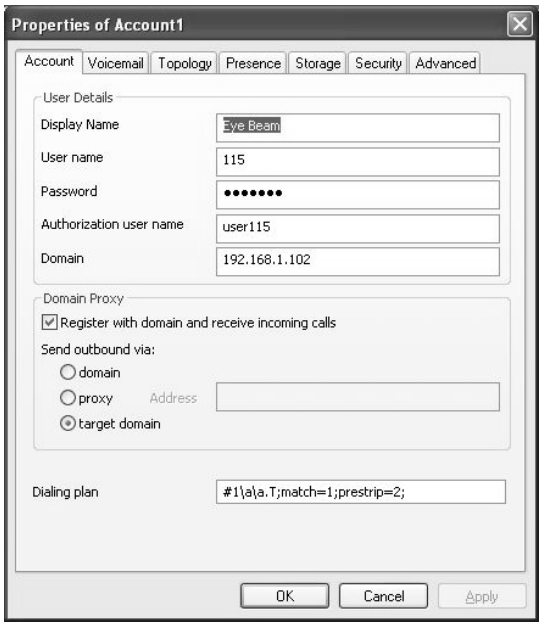
Configuring a new SIP account

The following procedure describes setting up the eyeBeam software to register on the network and to make itself available as a client for sending and receiving calls.

1. Install the eyeBeam software. The **SIP Accounts** window appears.



2. Click the **Add** button. The **Properties of Account1** window appears.



3. Enter the *User Details* in the *Account* tab.

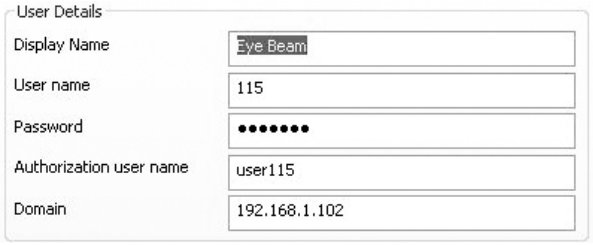
   a) Set the *Display Name* to the name for Caller ID.

   b) Set the *User name* to the local extension number.

   c) Set the *Password* to the *Password* set up in step 3 of *IP extension details area* on page 111.

   d) Set the *Authorization user name* to the *Username* set up in step 3 of *IP extension details area* on page 111.

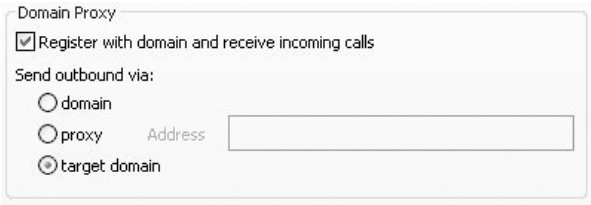
e) If setting up an internal IP extension, set *Domain* to the IP address of the TalkSwitch unit acting as local proxy. If setting up an external IP extension, set *Domain* to the public IP address or FQDN (Fully Qualified Domain Name) of the TalkSwitch system.



4. Set up the *Domain Proxy* area in the *Account* tab.

   a) Select the *Register with domain and receive incoming calls* checkbox.

   b) Select the *target domain* option.



5. Set up the *Voicemail* tab.

   a) Select *Check for voice mail*.

   b) Enter ** in the *Number to dial for checking voicemail* field.

   c) Enter *<*voicemail number*> in the *Number for sending calls to voicemail* field.

6. Click **OK** to save settings and close the **Properties of Account1** window.

7. Click **Close** to close the **SIP Accounts** window and enable the newly configured SIP account. Once closed, the phone display will show:

   - Discovering network…
   - Registering…
   - Ready, Your username is: [local extension number].

Updating eyeBeam software

The latest release of eyeBeam can be found at **www.counterpath.net/eyebeam.html**.

ADDING OTHER IP PHONES

Other IP phones with the G.711 codec may work with TalkSwitch but not all features may be supported. We strongly recommend you use only the IP phones supported by TalkSwitch.

If you connect an unsupported IP phone, select *Other IP phone* as the *Type of phone*. As TalkSwitch cannot enable features, or customizations of these phones, further configuration will be limited to the programmable options on the IP phone itself.

Refer to the specific IP phone user guide's details for information on configuration.

REMOTE EXTENSIONS

A remote extension reaches an external phone by automatically selecting a line from a hunt group and dialing the phone number. For example, a remote extension could reach an employee's cell phone or home phone, or a phone at a branch office.

A caller can connect to a remote extension through the auto attendant, or can be transferred to a remote extension by a call cascade. A user at a local extension can manually transfer a caller to a remote extension, or can dial a remote extension directly. If the remote extension is busy or unanswered, the system can route the call using the remote extension's call cascade.

For example, a caller reaches the auto attendant and dials a local extension. The user is not there, so the call is unanswered. The call cascade of the local extension can be configured to transfer unanswered calls to a remote extension. The remote extension can be configured to dial the user's cellular phone. This way the user is available outside the office.

There are three ways the TalkSwitch system can transfer calls to a remote extension:

- If the telephone line has the Transfer and Clear service available and activated, the TalkSwitch system can use the transfer and clear feature. When an outside caller is being routed to a remote extension, the TalkSwitch system directs the telephone company to put the caller on hold. It then uses the same line to reach the remote extension. The telephone company connects the outside caller to the remote extension, and then frees the line.
- If the telephone line has the 3-Way Calling/Conference service available and activated, the TalkSwitch system can use the same line connect feature. When an outside caller is being routed to a remote extension, the TalkSwitch system directs the telephone company to put the caller on hold. It then uses the same line to reach the remote extension. The telephone company connects the outside caller to the remote extension, but the line to the TalkSwitch unit remains occupied until the call is complete.
- If the telephone line doesn't have these services, and an outside caller is being routed to a remote extension, the TalkSwitch system puts the caller on hold. It then uses a second line to reach the remote extension. Both lines remain occupied until the call is complete.

If two lines are used to connect an outside caller to a remote extension, the user at the remote extension can:

- Place the call on hold by dialing **.
- Retrieve the call on hold by dialing **.
- Transfer the call by dialing ** + the extension number. The call can be transferred to a local extension, remote extension, ring group, or VoIP number within a TalkSwitch VoIP network.
- Transfer the call to a voice mailbox by dialing *** + the voice mailbox number.

The remote extensions can be configured to perform a blind transfer or a screened transfer. However a transfer to another remote extension is always a blind transfer to avoid tying up a third line.

If the system is configured to perform a blind transfer, it plays "Call transferred. Goodbye." and then hangs up. The call is transferred to the dialed extension and follows its call cascade if busy or unanswered.

If the system is configured to perform a screened transfer, the user is connected to the other extension and asks whether they want the call. If so, the user dials ** + **4** to complete the transfer. If the other person doesn't want the call, the user dials ** + **5** to cancel the transfer and return to the caller.

There are ten remote extensions per TalkSwitch unit, which are numbered as follows:

- TalkSwitch Unit 1 — 210 to 219
- TalkSwitch Unit 2 — 220 to 229
- TalkSwitch Unit 3 — 230 to 239
- TalkSwitch Unit 4 — 240 to 249

The **Remote Extensions** page allows you to set up a remote extension.

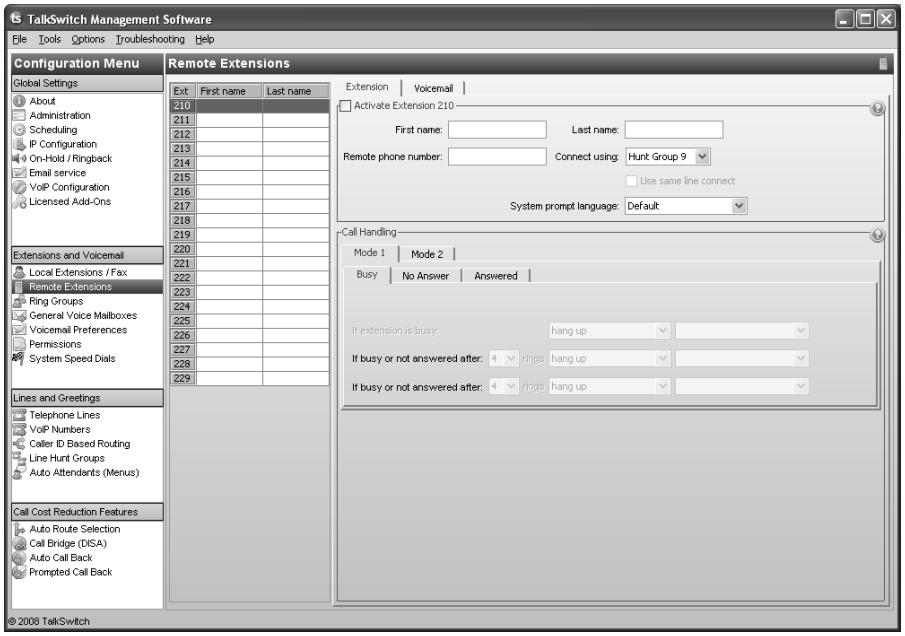
Remote extensions are designed to operate with local major telephone service providers. The feature may not function correctly with some telephone and mobile operator's networks, especially for international phone numbers and mobile phones roaming internationally.

Extension tab

The *Extension* tab allows you to set the name, phone number, hunt group, language, same line connect feature, and call cascade for each remote extension.

1. Select the **Remote Extensions** page.
2. Select a slot for the remote extension.

3. Select the *Extension* tab.



Activate extension

The *Activate Extension* area allows you to set the name, phone number, hunt group, language, and same line connect feature for each remote extension.

1. Select the *Activate Extension* checkbox.

2. Enter the user's *First name* and *Last name*. The names appear within the configuration windows. The *Last name* is used with the dial-by-name directory.

3. Enter the *Remote phone number*. Enter the number as you would normally dial it (i.e. without the hunt group number). You can enter digits 0–9, space, dash, comma, **#** and *****. A comma pauses dialing for two seconds.

   An administrator can change the phone number by phone. Enter command mode by either pressing **#** on an analog extension phone, or ***55#** on a TalkSwitch IP phone (note: other brands may use ***55 Send** or ***55 Dial**). Dial the system password + **#**, the remote extension number, and then **8** + the new phone number + **#**.

4. Select the hunt group in the *Connect using* list. The TalkSwitch unit will use a line from this hunt group to connect with the remote extension. We recommend the default *Hunt group 9* (or *Hunt Group 0* in certain regions) unless you have set up a different hunt group for calling remote extensions.

   An administrator can change the hunt group by phone. Enter command mode by either pressing **#** on an analog extension phone, or ***55#** on a TalkSwitch IP phone (note: other brands may use ***55 Send** or ***55 Dial**). Dial the system password + **#**, the remote extension number, and then **9** + the new hunt group + **#**.

5. Select the language heard by the user of the extension in the *System prompt language for this extension*.

6. If you have the 3-Way Calling/Conference service from the telephone company, you can select the *Use same line connect* checkbox. The same line connect feature will direct the telephone company to put the caller on hold, and will then use the same line to try the remote extension. Because the same line is used, the hunt group setting from Step 4 is ignored.

   If you do not select the *Use same line connect* checkbox, the TalkSwitch unit will put the caller on hold, and then use a second line to try the remote extension. The second line will be from the hunt group selected in Step 4.

   If you enable the same line connect feature, the user at the remote extension will not be able to transfer the call to another extension.

   When the same line connect feature is in use, the caller hears silence while they are on hold. This is because the caller is on hold at the telephone company and not at the TalkSwitch unit. If you want the caller to hear music on hold while their call is being forwarded to the remote extension, do not use the same line connect feature.

   The same line connect feature is not compatible with some telephone company lines.

   An administrator can change the *Use same line connect* setting by phone. Enter command mode by either pressing **#** on an analog extension phone, or ***55#** on a TalkSwitch IP phone (note: other brands may use ***55 Send** or ***55 Dial**). Dial the system password + **#**, the remote extension number, and then **9800#** to turn off same line connect, or **9801#** to turn on same line connect.

7. Set the type of transfer performed by users at remote extensions. This affects transfers to local extensions, ring groups, and VoIP numbers within a TalkSwitch VoIP network. Transfers to other remote extensions are always blind transfers. See *To remote extension tab* on page 184.

Transfer Preferences
Specify the options for transferring a caller:
To Local Extension
To Remote Extension
To Ring Group
From Auto Attendant
From Home Phone
If the extension is busy
Ring back the extension that transferred the call
Perform the Busy action
If the extension does not answer
Ring back the extension that transferred the call
Perform the No Answer action
Transfer and Clear
Clear telephone line after call transferred
Remote Transfer
When transferring calls from a remote extension:
Allow screening of calls
Perform a blind transfer
Use Defaults
OK
Cancel

   a) Choose **Options > Transfer Preferences**. The **Transfer Preferences** window appears.

   b) Select the *To Remote Extension* tab.

      - To allow screened transfers, select *Allow screening of calls*.
      - To allow blind transfers, select *Perform a blind transfer*. This is the default selection.

8. If you have the Transfer and Clear service from the telephone company, you can enable the transfer and clear feature. See *To remote extension tab* on page 184.

   a) Choose **Options > Transfer Preferences**. The **Transfer Preferences** window appears.

   b) Select the *To Remote Extension* tab.

   c) To enable the transfer and clear feature, select the *Clear telephone line after call transferred* checkbox.

About call cascades

See *About call cascades* on page 65.

Setting up call handling

The procedure for setting up a call cascade is similar to that described in *Local Extension* with the differences described below. See *Setting up call handling* on page 67.

- The **Remote Extensions** page does not have the *Do Not Disturb* tab.
- The *If Extension is busy* list in the *Busy* tab does not have the *invoke call waiting* or *queue at extension* options.
- The *When a call is answered* list in the *Answered* tab has the *play accept/reject prompt* option, in addition to the *stay connected* and *play caller's name first* options.

  If *play accept/reject prompt* is selected, the system will place the caller on hold and will then call the remote extension. When the user answers, the system will say: *"You have a forwarded call. To accept the call, press* **#**. *To reject the call, press* *****." If the user accepts the call, the system will route the call to the remote extension. If the user rejects the call, the system will try the next alternative in the remote extension's call cascade.

When setting the number of rings to wait for a remote extension to connect, it is very important to allow enough time for the telephone network to connect to the phone. For example, a call to a cell phone or PDA may take 12 seconds (i.e. 2 rings) just to connect.

In the following circumstances, the system will start by saying *"This is call cascade"*, instead of *"This is call forward"* or *"You have a forwarded call"*.

- An extension's call cascade uses *play caller's name first* or *play accept/reject prompt*, and
- The previous user answered but rejected the call, and
- The extension's call cascade is now attempting to transfer the call to a remote extension.

This notifies the user that the call was rejected. If the user also rejects the call, it will be routed according to the other extension's call cascade.

Voicemail tab

The procedure for setting up voicemail is identical to that described in *Local Extension/Fax*. See *Voicemail tab* on page 75.

Mailbox greeting

The procedure for setting up the mailbox greeting is identical to that described in *Local Extension/Fax*. See *Mailbox greeting* on page 76.

Caller options

The procedure for setting up the caller options is identical to that described in *Local Extension/Fax*. See *Caller options* on page 76.

Notification settings

The procedure for setting up the notification settings is identical to that described in *Local Extension/Fax*. See *Notification settings* on page 77.

Remote extensions and telephone company services

See *Appendix B: TalkSwitch and Telephone Company Calling Services* on page 267 for information about using the 3-Way Calling/Conference, Transfer and Clear, Prompted Ring Back, and Ringback Tones services with remote extensions.

RING GROUPS

A ring group is a group of local extensions that ring in unison. Local extensions and auto attendants can dial a ring group.

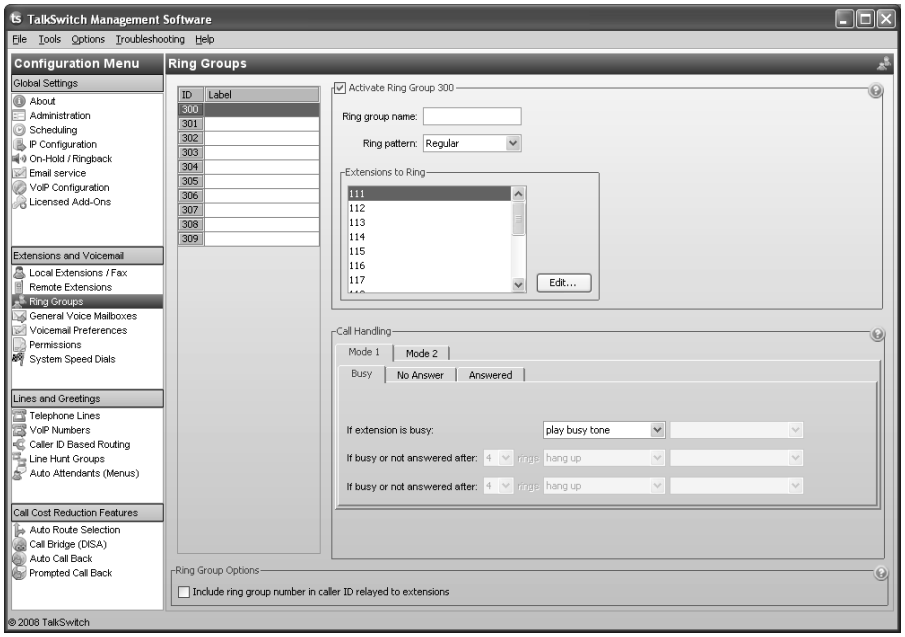
There are ten ring groups in the TalkSwitch system, which are numbered 300 to 309. By default, 300 is configured to ring all local extensions.

Ring groups have two main uses:

- A ring group can reach a group of employees. For example, ring group 301 can ring the sales group at extensions 111, 112, 113, and 114. When a customer calls the sales group, the first available salesperson answers for the group.
- A ring group with a different ring pattern can differentiate callers. For example, the president doesn't want to answer calls from the general public, but wants to be alerted when important colleagues are on the line. His local extension is 111, but you can configure ring group 305 to ring his phone with a different ring pattern. The president tells his colleagues to reach him at extension 305, which is his private extension. He can tell by the ring pattern if the caller dialed extension 111 or 305, and can react accordingly.

The **Ring Groups** page allows you to set up the name, ring pattern, local extensions, and call cascade for each ring group.

1. Select the **Ring Groups** page.
2. Select a slot for the ring group.



Activate ring group

The *Activate Ring Group* area allows you to set up the name, ring pattern, and local extensions for each ring group.

1. Select the *Activate Ring Group* checkbox.
2. Enter the *Ring group name*. The name appears within the configuration windows.

3. Select a *Ring pattern* to indicate the call is for the ring group. Choices are *Regular*, and *Alternate 1* to *Alternate 5*.

4. Click the **Edit** button. The **Set Ring Group Extensions** window appears.

5. Select the local extensions you want to ring in unison.

About call cascades

See *About call cascades* on page 65.

Setting up call handling

The procedure for setting up a call cascade is similar to that described in *Local Extension/Fax* with the differences described below. See *Setting up call handling* on page 67.

- The **Ring Groups** page does not have the *Do Not Disturb* tab.
- The *If Extension is busy* list in the *Busy* tab does not have the *invoke call waiting* option.
- The *queue at ring group* option replaces *queue call*, and places the call in the ring group's call queue.

Ring group options

The *Ring Group Options* area allows you to set up the system so users will know the caller is trying to contact the ring group, and not them personally.

1. Select the *Include ring group number* checkbox to display the ring group number with the caller ID information. For example, if John Smith is calling ring group 300, each extension in ring group 300 will show "300 - John Smith".

GENERAL VOICE MAILBOXES

General voicemail is not associated with any extension, but is for general use or for a group. By default, general voicemail is deactivated.

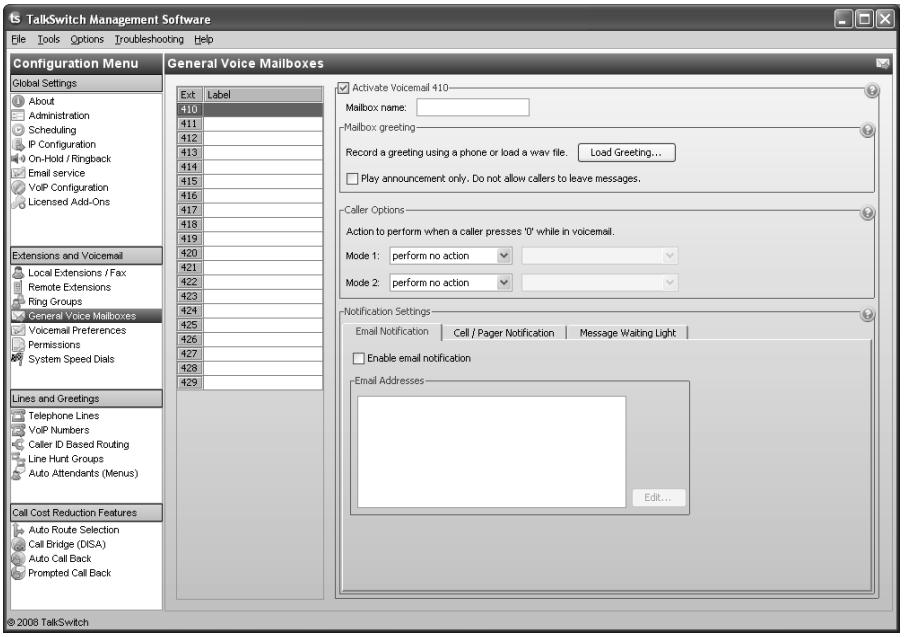
There are ten general voice mailboxes per TalkSwitch unit, which are numbered as follows:

- TalkSwitch Unit 1 — 410 to 419
- TalkSwitch Unit 2 — 420 to 429
- TalkSwitch Unit 3 — 430 to 439
- TalkSwitch Unit 4 — 440 to 449

The procedure for setting up general voicemail is similar to that described in *Local Extension/Fax*, except each general voice mailbox is deactivated by default. See *Voicemail tab* on page 75.

The **General Voice Mailboxes** page allows you to set up general voicemail.

1. Select the **General Voice Mailboxes** page.



Note: In the *Caller Options* area, the screen may refer to when a caller presses '9' depending on the region.

Mailbox greeting

The procedure for setting up the mailbox greeting is identical to that described in *Local Extension/Fax*. See *Mailbox greeting* on page 76.

Caller options

The procedure for setting up the caller options is identical to that described in *Local Extension/Fax*. See *Caller options* on page 76.

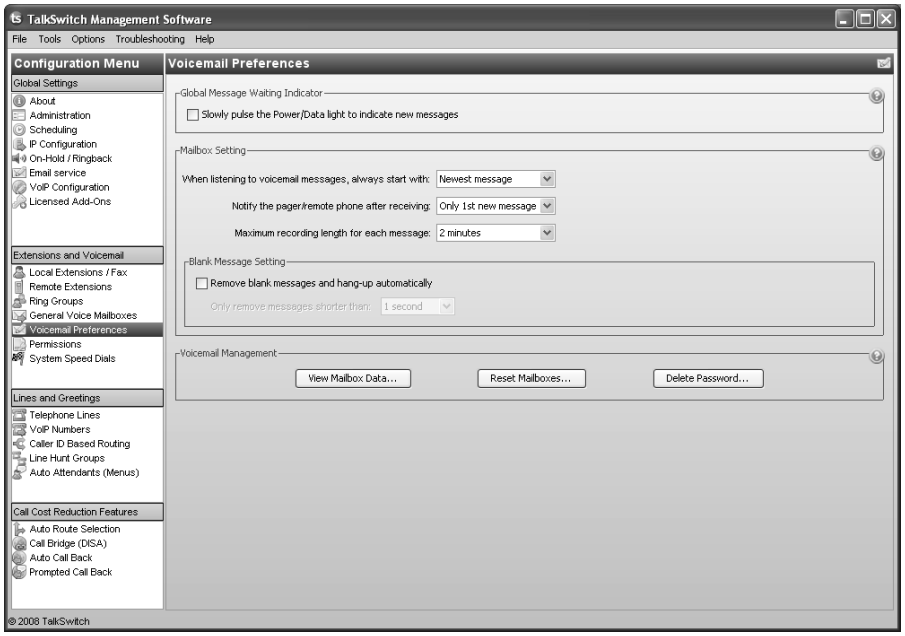
Notification settings

The procedure for setting up the notification settings is identical to that described in *Local Extension/Fax*. See *Notification settings* on page 77.

VOICEMAIL PREFERENCES

The **Voicemail Preferences** page allows you to set up the global message waiting indicator light and the voice mailbox settings. Any changes entered on this window affect all mailboxes. You can also view mailbox data, reset mailboxes and delete the password for a mailbox.

1. Select the **Voicemail Preferences** page.



Global message waiting indicator

The *Global Message Waiting Indicator* area allows you to have the Power/Data light on the TalkSwitch unit flash when there is a voicemail message in any mailbox.

1. Select the *Slowly pulse the Power/Data light to indicate new messages* checkbox to have the Power/Data light on the TalkSwitch unit flash when voicemail is present.

Mailbox setting

The *Mailbox Setting* area allows you to set up general mailbox settings.

1. Set the *When listening to voicemail messages, always start with* list to which message should be played first. Choices are *Newest message* or *Oldest message*.

2. Set the *Notify the pager/remote phone after receiving* list to how often the system should send notification. Choices are *Every new message* or *Only 1st new message*.

3. Set the *Maximum recording length for each message* list to how long a message can be. Choices range from *1 minute* to *8 minutes*.

4. Select the *Remove blank messages and hang up automatically* checkbox, if you want the TalkSwitch system to delete blank messages. A blank message occurs when the caller hangs up upon reaching your voicemail. If you have the caller ID service from your telephone company and choose to keep blank messages, you will be able to check the caller's number when they leave a blank message.

5. If you selected the *Remove blank messages and hang up automatically* checkbox, set the *Only remove messages shorter than* list. The TalkSwitch system will automatically remove messages that are shorter than the selected length. Choices range from *1 second* to *5 seconds*.

Voicemail management

The *Voicemail Management* area has buttons that allow you to view mailbox data, reset voice mailboxes, and delete the password for a mailbox.

View mailbox data

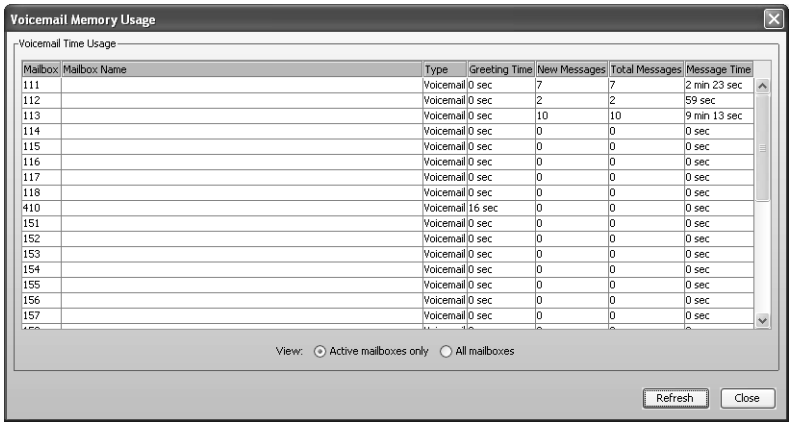
The **Voicemail Memory Usage** window shows how much recording time is taken up by voicemail greetings and messages.

The recording time for internal music on hold, voicemail, and the auto attendants is shared on the TalkSwitch unit. You can expand the built-in memory using memory cards.

Note that the system allows up to 99 voicemail messages per mailbox. Once a mailbox fills up, callers won't be able to leave voicemail messages for that user. Therefore users should delete voicemail messages before the mailbox fills up.

1. Click **View Mailbox Data**. The **Voicemail Memory Usage** window appears.

   Alternatively, you can choose **Tools > Voicemail Manager > Mailbox Status**.


Voicemail Memory Usage
Voicemail Time Usage
Mailbox
Mailbox Name
Type
Greeting Time
New Messages
Total Messages
Message Time
111 Voicemail 0 sec 7 7 2 min 23 sec
112 Voicemail 0 sec 2 2 59 sec
113 Voicemail 0 sec 10 10 9 min 13 sec
114 Voicemail 0 sec 0 0 0 sec
115 Voicemail 0 sec 0 0 0 sec
116 Voicemail 0 sec 0 0 0 sec
117 Voicemail 0 sec 0 0 0 sec
118 Voicemail 0 sec 0 0 0 sec
410 Voicemail 16 sec 0 0 0 sec
151 Voicemail 0 sec 0 0 0 sec
152 Voicemail 0 sec 0 0 0 sec
153 Voicemail 0 sec 0 0 0 sec
154 Voicemail 0 sec 0 0 0 sec
155 Voicemail 0 sec 0 0 0 sec
156 Voicemail 0 sec 0 0 0 sec
157 Voicemail 0 sec 0 0 0 sec
View: Active mailboxes only All mailboxes
Refresh
Close

2. Select *Active mailboxes only* to conceal the deactivated mailboxes, or select *All mailboxes* to show the deactivated mailboxes as well.

Reset mailboxes

Resetting mailboxes deletes greetings, messages and passwords from the selected local extension, remote extension and general mailboxes. You can optionally delete the names recorded for the dial-by-name directory from these mailboxes as well.

1. Click **Reset Mailboxes**. The **Reset Mailboxes** window appears.

Alternatively, you can choose **Tools > Voicemail Manager > Reset Mailboxes**.

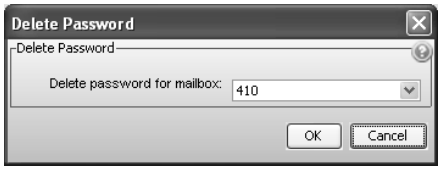
2. Select one or more mailboxes to delete their greetings, messages and passwords.
3. Select the *Delete name from dial-by-name directory* checkbox to delete the names recorded for the dial-by-name directory from these mailboxes.
4. Click **Reset Mailboxes**. The greetings, messages and passwords (and optionally names for the dial-by-name directory) are deleted from the selected mailboxes.

Delete password

You can delete the password from the selected local extension, remote extension or general mailbox.

1. Click **Delete Password**. The **Delete Password** window appears.

   Alternatively, you can choose **Tools > Voicemail Manager > Delete Mailbox Password**.



2. Select the mailbox to delete its password.

PERMISSIONS

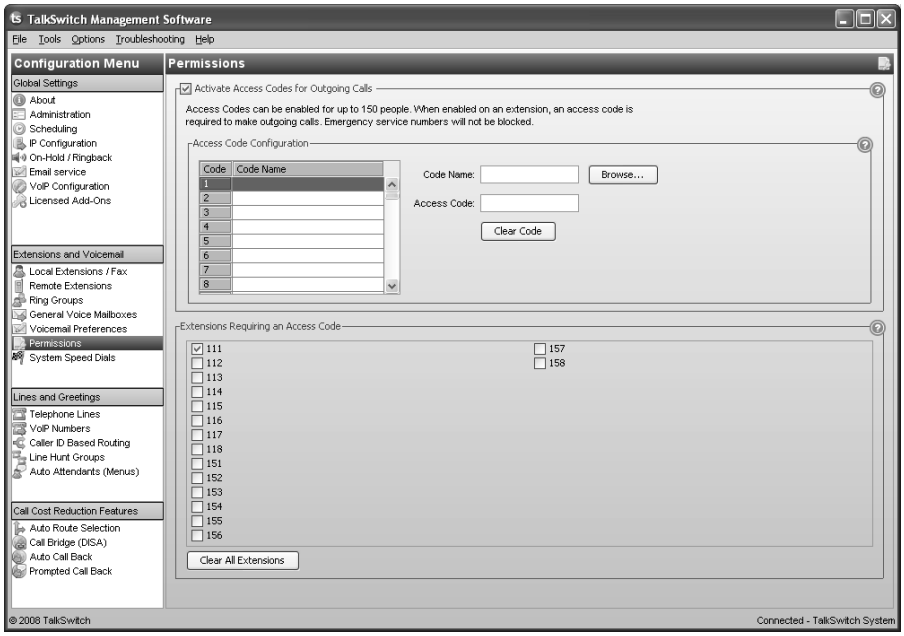
A local extension can be restricted. If so, it will require an access code before an outside phone number can be dialed. For example, you can configure the lobby phone to require an access code. The receptionist can provide an access code to visitors, as required. However an access code is not required to answer a call, dial an emergency service number or place an intercom call.

To use an access code, pick up a local extension and dial the hunt group. If an access code is required, the TalkSwitch system will prompt you for the access code. Enter the access code, followed by **#**. If the access code is accepted, the TalkSwitch unit makes a telephone line available.

Enable password protection when access codes are used. Change the system password frequently to prevent unauthorized users from making calls or changing the configuration. See *Administration* on page 27.

The **Permissions** page allows you to set up access codes for up to 150 people. You can restrict local extensions, by selecting those that require an access code to make an outside call.

1. Select the **Permissions** page.



Activate access codes

The *Activate Access Codes for Outgoing Calls* area allows you to enable the access code feature. You can then set up access codes for up to 150 people. You can also restrict local extensions, by selecting those that require an access code to make an outside call.

1. Select the *Activate Access Codes for Outgoing Calls* checkbox. The **Permissions** page allows you to enter access codes and restrict local extensions.

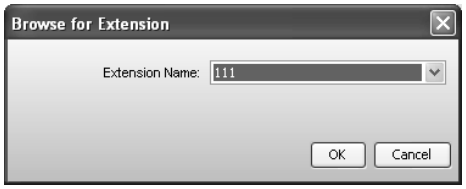
Access code configuration

The *Access Code Configuration* area allows you to add and remove access codes.

Adding an access code

1. Select a code.
2. Enter a *Code name*. This is typically the name of the user, or the name of the office where the extension is located.

   Alternatively, you can click the **Browse** button to display the **Browse for Extension** window. It allows you to select an extension number and the associated first name and last name as the code name.

   If a call is made from a restricted local extension, the call detail record (CDR) will display the code name associated with the access code, and not the access code itself.
3. Enter an *Access code*. This is a numeric code that can be dialed using a telephone keypad. Each access code works with all restricted extensions.

Removing an access code

1. Select the access code.
2. Click **Clear Code**.

Extensions requiring an access code

The *Extensions Requiring an Access Code* area allows you to restrict selected extensions, or clear all restrictions. Restricted local extensions will require an access code to make an outgoing call.

Restricting extensions

1. Select the checkboxes of the local extensions to restrict them.

Clearing restrictions

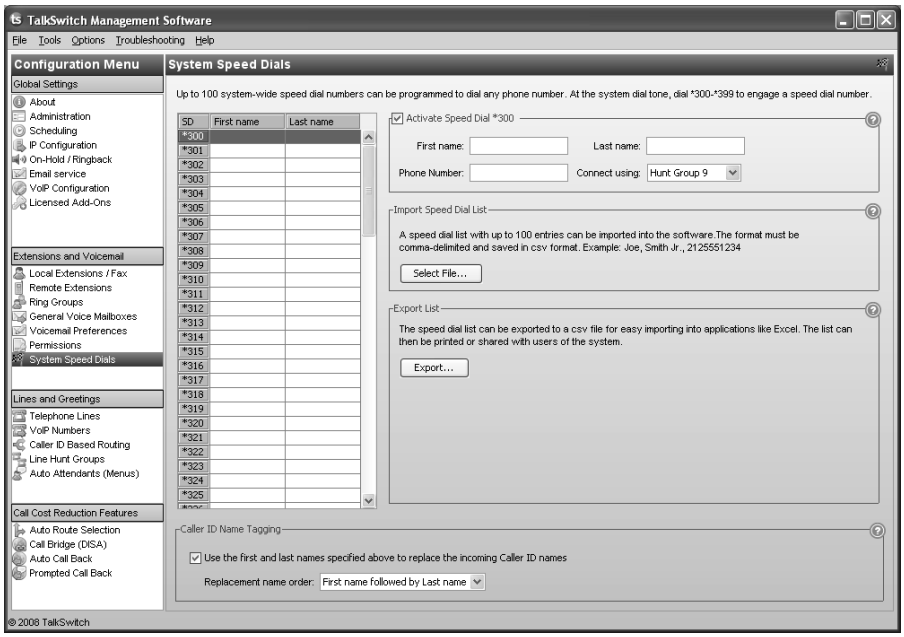
1. Click **Clear All Extensions** to make all local extensions unrestricted.

SYSTEM SPEED DIALS

System speed dials allow the user to quickly dial an outside phone number from any local extension by dialing a speed dial number ranging from ***300** to ***399**. Because the speed dial numbers are maintained within the TalkSwitch system, they don't need to be programmed into each individual phone. If a new client is added or an existing client's phone number changes, you can add or modify the speed dial number, and all users will have immediate access to the new phone number.

The **System Speed Dials** page allows you to define up to 100 speed dial numbers. Each speed dial number includes the first name, last name, phone number and hunt group. You can create a .csv file that contains speed dial numbers, and then import them into the system. You can also export the speed dial numbers to a .csv file. Caller ID name tagging can be enabled, which will substitute the caller ID name with the system speed dial name.

1. Select the **System Speed Dials** page.


TalkSwitch Management Software
File Tools Options Troubleshooting Help
Configuration Menu
Global Settings
About
Administration
Scheduling
IP Configuration
On-Hold / Ringback
Email service
VoIP Configuration
Licensed Add-Ons
Extensions and Voicemail
Local Extensions / Fax
Remote Extensions
Ring Groups
General Voice Mailboxes
Voicemail Preferences
Permissions
System Speed Dials
Lines and Greetings
Telephone Lines
VoIP Numbers
Caller ID Based Routing
Line Hunt Groups
Auto Attendants (Menus)
Call Cost Reduction Features
Auto Route Selection
Call Bridge (DISA)
Auto Call Back
Prompted Call Back
System Speed Dials
Up to 100 system-wide speed dial numbers can be programmed to dial any phone number. At the system dial tone, dial *300-*399 to engage a speed dial number.
SD
First name
Last name
Activate Speed Dial *300
First name:
Last name:
Phone Number:
Connect using:
Hunt Group 9
Import Speed Dial List
A speed dial list with up to 100 entries can be imported into the software.The format must be comma-delimited and saved in csv format. Example: Joe, Smith Jr., 2125551234
Select File...
Export List
The speed dial list can be exported to a csv file for easy importing into applications like Excel. The list can then be printed or shared with users of the system.
Export...
Caller ID Name Tagging
Use the first and last names specified above to replace the incoming Caller ID names
Replacement name order:
First name followed by Last name
© 2008 TalkSwitch

Activate speed dial

The *Activate Speed Dial* area allows you to define up to 100 speed dial numbers. Each speed dial number includes the first name, last name, phone number and hunt group.

If the user dials an inactive speed dial number, they will hear an error message.

1. Select a speed dial number.
2. Select the *Activate Speed Dial* checkbox.
3. Enter the *First name*.
4. Enter the *Last name*.
5. Enter the *Phone number*. Valid characters are **0**–**9**, **-**, **,** (pauses dialing for two seconds), **w** (pauses dialing until a dial tone is detected), ***** and **#**. The phone number can be up to 19 characters long.

   The phone number can be a regular phone number or a VoIP number from a service provider.

If the phone number is a regular phone number, it can optionally include a pause followed by an extension number. For example, the phone number is 555-1212 and the desired extension is 123. The *Phone number* can be set to **555-1212,123**.

NOTE: The phone number cannot include an extension number if it is a VoIP number from a service provider.

If auto route selection blocks the phone number, the phone number will not be blocked if a user dials the speed dial number.

6. Select the hunt group. Choices are the active hunt groups.

   If auto route selection or permissions block the hunt group, the hunt group will not be blocked if a user dials the speed dial number.

   If all lines in the hunt group are busy, but an overflow hunt group is set up, the system will try the overflow hunt group.

Creating a speed dial list

A speed dial list is a .csv file that contains speed dial numbers. You can create the speed dial list using a text editor like Microsoft Notepad, or with a spreadsheet program like Microsoft Excel. You can then import the speed dial list into the TalkSwitch system.

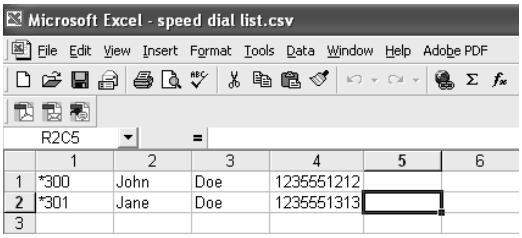
Including the speed dial number

Each entry in the file can include the speed dial number. It requires the following information: *speed dial number, first name, last name, and phone number. When you import the file, and an entry has the same speed dial number as an existing speed dial number, the entry will override and replace the existing speed dial number.

If using a text editor (e.g. Microsoft Notepad), surround each item with quotes. For example:

```
"*300","John","Doe","1235551212"
"*301","Jane","Doe","1235551313"
```

If using a spreadsheet program (e.g. Microsoft Excel), omit the quotes. For example:


Microsoft Excel - speed dial list.csv
File Edit View Insert Format Tools Data Window Help Adobe PDF
R2C5
1 2 3 4 5 6
1 *300 John Doe 1235551212
2 *301 Jane Doe 1235551313
3

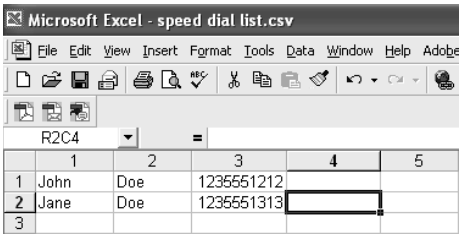
Omitting the speed dial number

Each entry in the file can omit the speed dial number. It requires the following information: first name, last name, and phone number. When you import the file, the system will automatically add the entries to the unused speed dial numbers, starting from the last provisioned speed dial number.

If using a text editor (e.g. Microsoft Notepad), surround each item with quotes. For example:

```
"John","Doe","1235551212"
"Jane","Doe","1235551313"
```

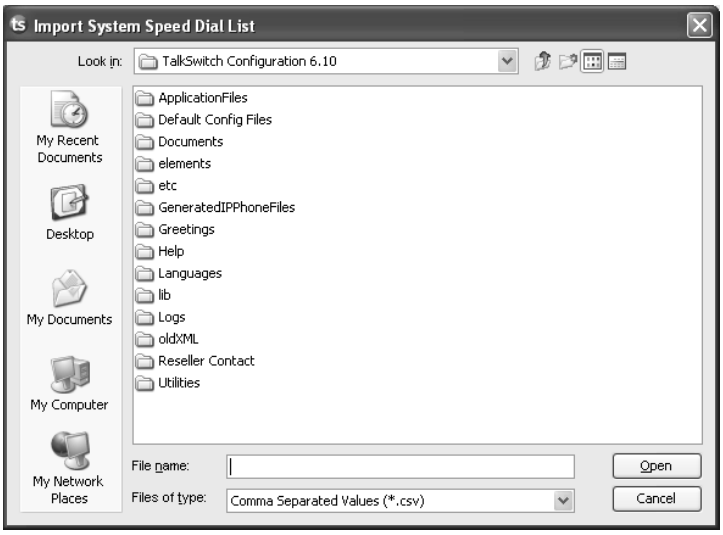
If using a spreadsheet program (e.g. Microsoft Excel), omit the quotes. For example:


Microsoft Excel - speed dial list.csv
File Edit View Insert Format Tools Data Window Help Adobe
R2C4
=
1 2 3 4 5
1 John Doe 1235551212
2 Jane Doe 1235551313
3

Import system speed dial list

The *Import System Speed Dial List* area allows you to import speed dial numbers from a .csv file. The imported speed dial numbers will override or be added to the existing speed dial numbers. Imported speed dial numbers will use the first available hunt group (i.e. 9, or 0 in some regions).

1. Click **Select File**. The **Import System Speed Dial List** window appears.



2. Select the .csv file, and then click **Open**. A progress bar appears.
3. Click **Close**.
4. Choose **File > Save** to save the imported speed dial list to the TalkSwitch unit.

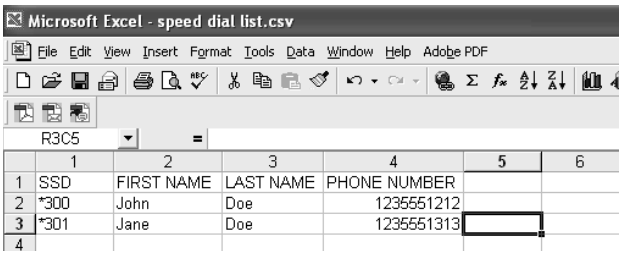
Export system speed dial list

The *Export System Speed Dial List* area allows you to export the speed dial numbers to a .csv file. You can then share the .csv file as required, or print it for your users. The first line of the file is a title line, and subsequent lines contain entries. Each entry in the file has the following information: *speed dial number, first name, last name, and phone number.

If you open the .csv file in a text editor (e.g. Microsoft Notepad), each item will be surrounded by quotes. For example:

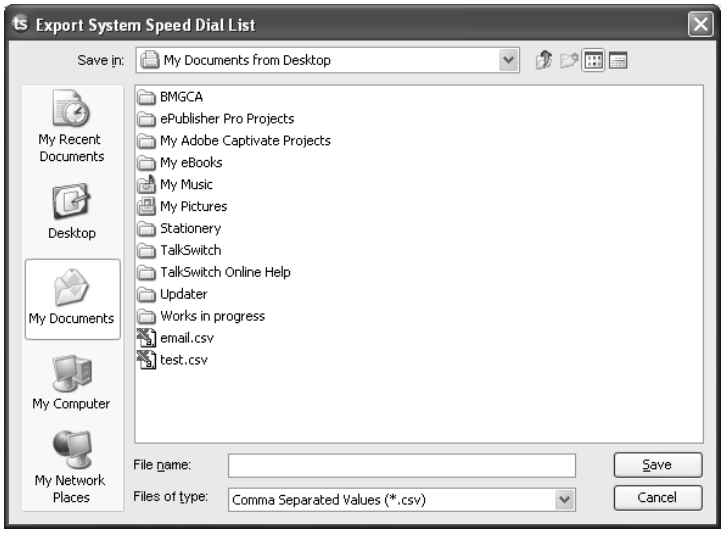
```
“SSD”,”FIRST NAME”,”LAST NAME”,”PHONE NUMBER”
“*300”,”John”,”Doe”,”1235551212”
“*301”,”Jane”,”Doe”,”1235551313”
```

If you open the .csv file in a spreadsheet program (e.g. Microsoft Excel), the quotes will be omitted. For example:


Microsoft Excel - speed dial list.csv
File Edit View Insert Format Tools Data Window Help Adobe PDF
R3C5
1 2 3 4 5 6
1 SSD FIRST NAME LAST NAME PHONE NUMBER
2 *300 John Doe 1235551212
3 *301 Jane Doe 1235551313
4

NOTE: If the phone number has more than 15 digits and does not have any special characters (**-**, **,**, **w**, ***** or **#**), Microsoft Excel will convert it to scientific notation. Do not attempt to import a .csv file that contains a phone number in scientific notation.

1. Click **Export**. The **Export System Speed Dial List** window appears.



2. Enter the name of the .csv file, and then click **Save**. A progress bar appears.

3. Click **Close**.

Caller ID name tagging

The *Caller ID Name Tagging* area allows you to enable the substitution of the caller ID name with the name from the **System Speed Dials** page.

Caller ID name tagging compares the phone number from the caller ID of an incoming call with the phone numbers from the **System Speed Dials** page. If a matching phone number is present, the system will replace the name from the caller ID with the first name and last name from the **System Speed Dials** page. These names will appear on the extension instead of the caller ID name.

For more information, see *Caller ID (or CLID) Based Routing* on page 145.

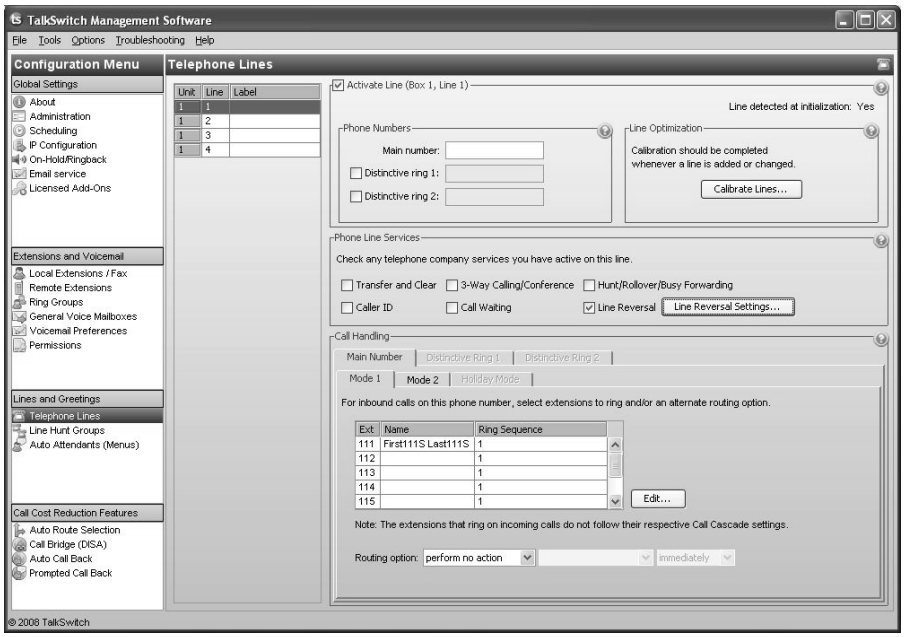
1. To enable caller ID name tagging of the entries within the **System Speed Dials** page, select the *Use the first and last names specified above to replace the incoming Caller ID names* check box.

2. Select the order for the names. Choices are *First name followed by Last name*, and *Last name followed by First name*.

TELEPHONE LINES

The **Telephone Lines** page allows you to set up the telephone numbers, telephone company services, and call handling for each telephone line. You can also calibrate the TalkSwitch unit to match the telephone lines.

1. Select the **Telephone Lines** page.



Activate line

The *Activate Line* area allows you to activate a telephone line. All telephone lines are activated by default.

"Line detected at initialization: Yes" appears if the telephone line was connected when you started the TalkSwitch management software. "Line detected at initialization: No" appears if the telephone line was not connected.

1. Select a telephone line.

2. If necessary, select the *Activate line* checkbox.

Phone numbers

The *Phone Numbers* area allows you to enter the main and distinctive ring phone numbers.

The Distinctive Ring service provides one or two additional phone numbers to ring the same telephone line. Each phone number has a different ring pattern. You can set up separate call handling for the main number, distinctive ring 1 number, and distinctive ring 2 number.

- Main number corresponds to the normal ring pattern.
- Distinctive ring 1 corresponds to a double ring pattern.
- Distinctive ring 2 corresponds to a triple ring pattern.

1. Enter the phone number of the telephone line in the *Main number* box.
2. If the telephone line has the Distinctive Ring service, and you want to set up different call handling for the distinctive ring numbers:

   a) Select the *Distinctive ring 1* and *Distinctive ring 2* checkboxes.

   b) Enter the distinctive ring numbers.

3. If the telephone line has the Distinctive Ring service, but you want the same call handling for each distinctive ring number, clear the *Distinctive ring 1* and *Distinctive ring 2* check boxes.

Phone line services

The *Phone Line Services* area allows you to select the telephone company services that are active on the telephone line. See *Appendix B: TalkSwitch and Telephone Company Calling Services* on page 267 for a description of these services and how the TalkSwitch unit uses them.

- *Transfer and Clear* allows the TalkSwitch unit to release the line after transferring a call from an outside caller to a remote extension. You can enable this feature if the Transfer and Clear service will allow a call to remain in progress between the outside caller and the remote extension, after the TalkSwitch unit hangs up. Enabling this feature without the service will result in the caller being disconnected when their call is transferred to a remote extension. To enable the feature, select the *Clear telephone line after call transferred* checkbox in the **Transfer Preferences** window. See *To remote extension tab* on page 184.
- *3-Way Calling/Conference* allows the same line connect feature to work with the line's 3-Way Calling/Conference service for call bridge (DISA) calls, and with incoming calls that are routed to remote extensions.
- *Hunt/Rollover/Busy Forwarding* indicates the line has a hunting facility or call forward on busy service. If an incoming call reaches a busy line, it will automatically try another line.
- *Caller ID* allows the TalkSwitch unit to forward caller ID information to the local extensions. The TalkSwitch unit can also use the time information provided by caller ID to update its internal clock.
- *Call Waiting* sounds a tone if an incoming call reaches a busy line. Call waiting is not recommended, since the TalkSwitch system cannot answer the second call. If your installation has more than one line consider using the telephone company's Hunt/Rollover service. The TalkSwitch system has call waiting built-in, and does not require the Call Waiting service from the telephone company.

When the Call Waiting service checkbox is enabled for an external telephone line, a user can pick up a waiting call on that line by pressing **Flash** or **Recall** then **89** on an analog phone, or **Xfer/Transfer/TRNF** then **89** on an IP phone.

- *Line Reversal* is used in some regions to exchange signals between the telephone company and your telephone equipment. If you select the *Line Reversal* checkbox, click the **Line Reversal Settings** button to select the reversal patterns used by your telephone company.

  Note that line reversal configuration is not offered in the United States or Canada.

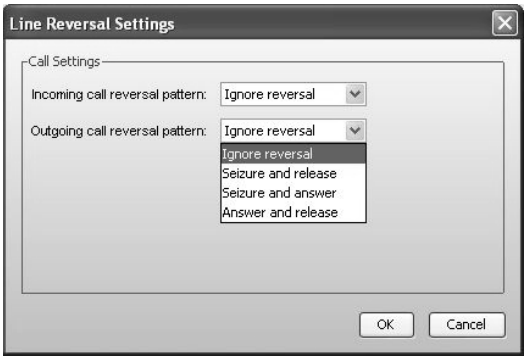
Line reversal settings

Line reversal is used in some regions to exchange signals between the telephone company and your telephone equipment. Line reversal can improve performance and lower costs on busy telephone lines. Check with your telephone company to see whether line reversal is used, and when. For example, the TalkSwitch unit can send a signal when it:

- Seizes the line. The TalkSwitch unit seizes a line when a user dials a hunt group number.
- Answers the line.
- Releases the line. The TalkSwitch unit releases the line when the user hangs up.

Note that line reversal configuration is not offered in the United States or Canada.

1. If line reversal is used, click **Line Reversal Settings**. The **Line Reversal Settings** window appears.



2. Select the reversal patterns specified by your telephone company for incoming and outgoing calls. The reversal patterns include:

   - *Ignore reversal* — Do not send or respond to line reversal signals.
   - *Seizure and release* — Send a signal when a line is seized, and again when the line is released.
   - *Seizure and answer* — Send a signal when a line is seized, and again when the line is answered.
   - *Answer and release* — Send a signal when a line is answered, and again when the line is released.

Call handling

The *Call Handling* area allows you to set up call handling for each telephone number of each telephone line for Mode 1, Mode 2 and Holiday Mode. The telephone line can ring selected local extensions in sequence, and then play an auto attendant or go to a voice mailbox. Alternatively, it can immediately play an auto attendant or go to a voice mailbox without ringing the extensions.

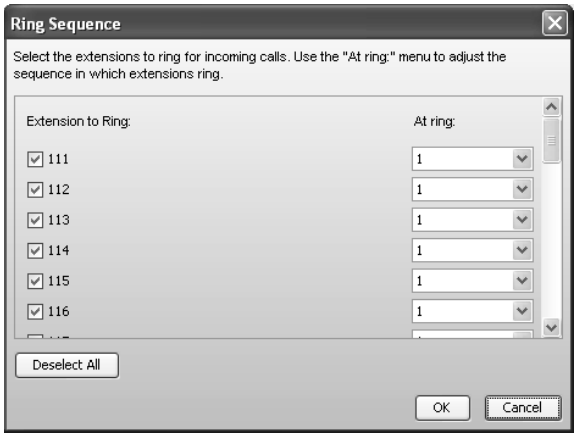
For example, the system can ring the receptionist when Mode 1 is active during the day. If the receptionist doesn't answer, it can start ringing other users as well. If there is still no answer, it can play an auto attendant. The auto attendant provides the dial-by-name directory, and allows the caller to dial an extension. A voice mailbox can immediately answer the telephone line when Mode 2 is active at night or on weekends. Another auto attendant can immediately route the call to a remote extension during Holiday Mode.

1. Select a telephone number tab. Choices are *Main number*, *Distinctive Ring 1* and *Distinctive Ring 2*.

2. Select the mode tab. Choices are *Mode 1*, *Mode 2* and *Holiday Mode*.

3. Set up the ring sequence. The ring sequence determines when each local extension will start ringing. The extensions will not use their call cascades during the ring sequence. The system transfers the call to the first extension that is answered.

   a) Click **Edit**. The **Ring Sequence** window appears.



   b) Select the local extensions to ring. All extensions are selected by default.

   c) For each selected extension, set when the local extension will start ringing. All extensions ring after the first ring by default.

      If you assign a different value to each extension, the extension with the lowest number of rings will ring first. The other extensions will start ringing as the number of rings goes up. For example, the system can ring the receptionist first, and then ring other users if the receptionist doesn't answer.

4. Set up call handling to route an unanswered call to an auto attendant or voicemail.

   a) Select the call handling action. Choices are:

      - *perform no action* — The system continues to ring the extensions. A generic auto attendant answers after 15 rings. It allows an authorized caller to make configuration changes, access voicemail or dial extensions.
      - *go to auto attendant* — Plays the selected auto attendant.
      - *go to voicemail* — Accesses the selected voice mailbox.
      - *play announcement* — Plays the selected announcement.

   b) If you selected *go to auto attendant*, select the auto attendant you wish to play in the selected mode. You can assign the same auto attendant to multiple telephone lines. Alternatively, if different organizations share the TalkSwitch system, a different auto attendant can be assigned to each telephone line.

   c) If you selected *go to voicemail* or *play announcement*, select the voice mailbox. You can select a local extension mailbox, remote extension mailbox, or general voice mailbox.

   d) Select when the system will perform call handling. Choices are:

      - *immediately* — The system will immediately perform call handling without using the ring sequence. Selecting *immediately* will disable the **Edit** button and the other ring sequence controls.
      - *after 1 ring* to *after 9 rings* — The system will follow the ring sequence first, and then perform call handling if no one answers.

Line optimization

For the best call audio quality and volume levels, your TalkSwitch system parameters must be matched to your telephone lines. The *Line Optimization* area offers two calibration methods to match your telephone lines.

The QUICK method finds a match for each phone line from a set of standard values. It provides a good match under typical phone line conditions. This method can be used without taking your TalkSwitch system out of service.

The OPTIMIZED method measures each phone line and creates custom parameters for the closest match. This method requires that your PC be connected on the same LAN as your TalkSwitch unit(s), and requires that your TalkSwitch system be out of service during the optimization process.

Both methods are described in detail in the **Calibrate Lines** window.

You should calibrate the lines:

- after adding or changing a telephone line.
- after connecting a line to equipment other than a telephone company line.
- if poor audio performance is encountered.

1. Click **Calibrate Lines**. The **Calibrate Lines** window appears.

Calibrate Lines...
Your telephone lines have been previously calibrated but are not optimized.
There are two methods of calibration: OPTIMIZED and QUICK.
The OPTIMIZED method:
requires that your PC be connected on the same LAN as your TalkSwitch unit(s)
provides the best match between your TalkSwitch and your telephone lines
is performed using a separate TalkSwitch Line Optimizer program (included in TalkSwitch software installation)
requires that your TalkSwitch system be out of service while the optimization program is running.
Based on your system's active lines, the OPTIMIZED method will take approximately 4 minutes.
The QUICK method:
provides a good match under typical phone line conditions
can be done without shutting down your telephone system
Based on your system's active lines, the QUICK method will take approximately 2 minutes.
Select one of the calibration methods and click a button below to continue.
QUICK
OPTIMIZED
Proceed
Cancel

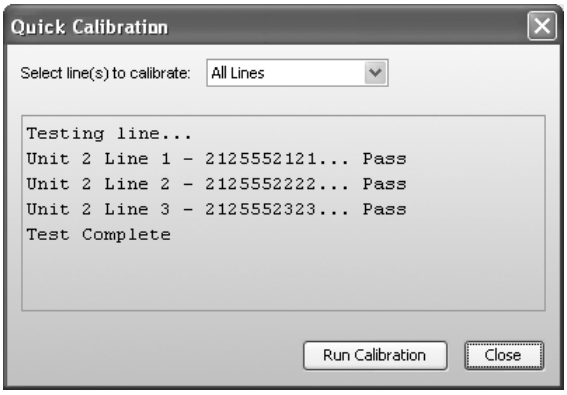
The calibration status of your system is displayed at the top of the window next to the information icon. You may be informed that:

- one or more of your telephone lines requires calibration.
- your telephone lines have been previously calibrated, but are not optimized.
- your telephone lines have been previously optimized and may not require further matching.

Select one of the calibration methods by clicking one of the radio buttons at left and click **Proceed** to continue.

QUICK method

A **QUICK Calibration** window will appear. Use the pull-down menu to select the lines to calibrate, and click **Run Calibration**. The software displays progress information throughout the calibration.



When **Test Complete** appears, click **Close**.

OPTIMIZED method

The OPTIMIZED method requires that your PC and your TalkSwitch unit(s) be on the same LAN.

You will be prompted to terminate your TalkSwitch configuration session before launching the TalkSwitch Line Optimizer program.

The TalkSwitch Line Optimizer should launch upon exit of the management software.

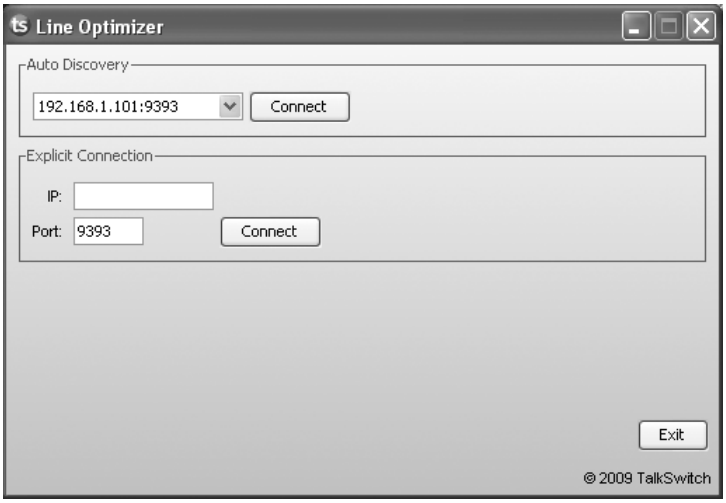
You can also launch this program from the start menu: **Programs > TalkSwitch 6.12 > Line Optimizer**.

If Windows provides a security alert when launching the Line Optimizer program, click **Ignore** or **Unblock** to proceed.

TalkSwitch Line Optimizer

The TalkSwitch Line Optimizer program automatically discovers TalkSwitch units on the same LAN that your PC is connected to. Use the pull-down menu to select the unit you want to optimize lines for. Click **Connect**.

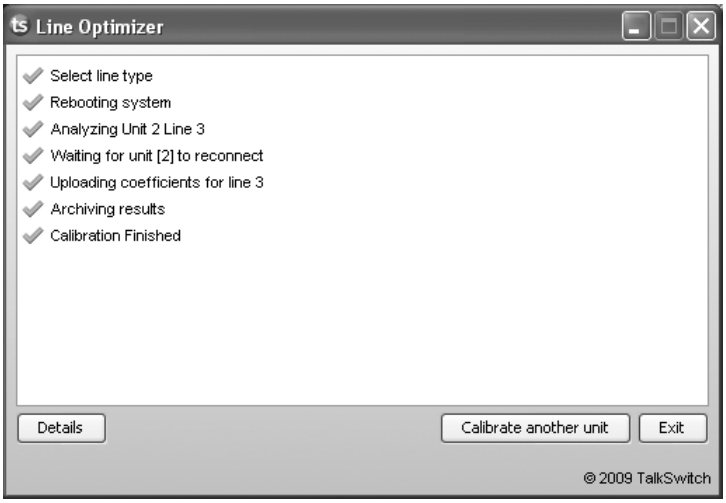
You can also enter the IP address of the desired TalkSwitch unit in the *IP* field in the *Explicit Connection* area. Specify the *Port* if necessary. The default port is *9393*.



Use the checkboxes to select the line(s) you want to optimize for each unit, and click **Calibrate...**

You will be prompted to allow a reboot of your TalkSwitch system. Click **Yes**.

As your TalkSwitch system reboots and proceeds with optimization, the Line Optimizer displays progress information.



When the program reports **Calibration Finished**, you can perform another calibration or exit the Line Optimizer.

VOIP NUMBERS

A VoIP number is like a telephone number. It allows a caller to dial a TalkSwitch system across the TalkSwitch VoIP network or through a VoIP service provider.

A VoIP number can be three or more digits long. Each VoIP number must be unique (i.e. only used at one location). The administrator for all locations needs to assign and keep track of the VoIP numbers within the TalkSwitch VoIP network.

VoIP numbers 250 to 299 permit direct dialing from any extension, auto attendant or call cascade to any extension, auto attendant, announcement or voice mailbox in the TalkSwitch VoIP network. No hunt group is required. Longer VoIP numbers can also be used, but the user must first dial a hunt group number.

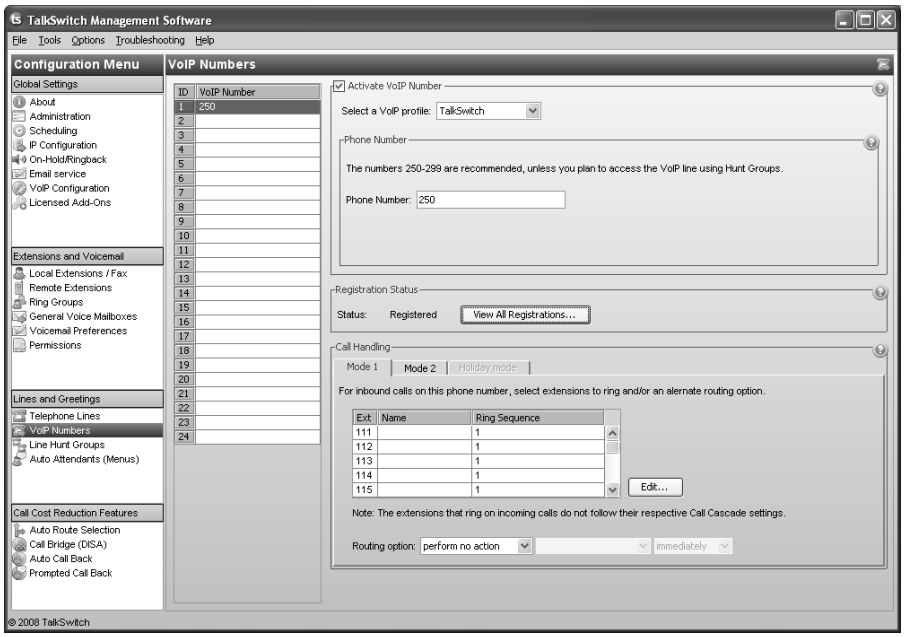
You can set up unique call handling for each VoIP number. Call handling for VoIP numbers is identical to call handling for telephone numbers. You can set up call handling for Mode 1, Mode 2 and Holiday Mode.

If a VoIP number is called, the system can ring selected local extensions in sequence, and then play an auto attendant or announcement, or go to a voice mailbox. Alternatively, it can immediately play an auto attendant or announcement, or go to a voice mailbox without ringing the extensions. For example, you can set up a general VoIP number that reaches the main auto attendant, and other VoIP numbers that are routed to specific extensions.

If you are using a TalkSwitch-certified VoIP service provider, visit the support section at **http://global.talkswitch.com** to access the service configuration guide for your VoIP service provider.

The **VoIP Numbers** page allows you to set up VoIP numbers. It is only present with a VoIP-enabled TalkSwitch unit.

1. Select the **VoIP Numbers** page.



Activate VoIP number

The *Activate VoIP Number* area allows you to activate a VoIP number and select its profile.

1. Select a VoIP number.
2. Select the *Activate VoIP Number* checkbox.
3. Select the profile for the VoIP number. Choices are:
    - TalkSwitch — The VoIP number will use the TalkSwitch VoIP network. The *Phone Number* area will contain the *Phone number* box.
    - Service provider (1 to 4) — The VoIP number will use the service provider. The *Phone Number* area will contain the *Country Code* box, *City or Area Code* box and *Number* box, and the *User Name and Password* area will be enabled.

Phone number

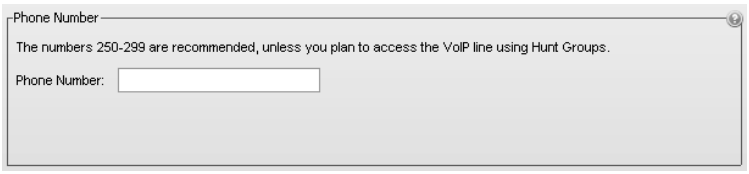
The *Phone Number* area allows you to enter the VoIP number.

TalkSwitch profile

If the TalkSwitch profile is used, a VoIP number can be three or more digits long. Each VoIP number must be unique (i.e. only used at one location).

VoIP numbers 250 to 299 permit direct dialing from any extension, auto attendant or call cascade to any extension, auto attendant, announcement or voice mailbox in the TalkSwitch VoIP network. Longer VoIP numbers can also be used, but the user must prefix them with a hunt group number.

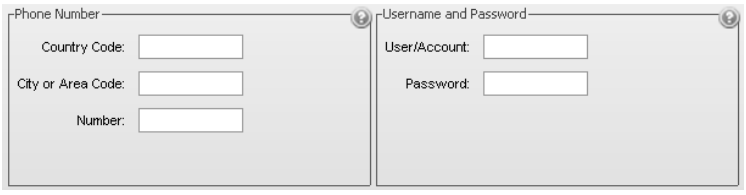
1. Enter the VoIP number into the *Phone number* box.



Service provider profile

If a service provider profile is used, use the country code, area code and phone number, as provided by the service provider.

1. Enter the *Country Code*, if required.
2. Enter the *City or Area Code*.
3. Enter the *Number*.



Username and password

If a service provider profile is used, the *Username and Password* area allows you to enter the authentication information required to access the service provider's SIP server.

1. Enter the *User/Account*, as provided by the service provider.
2. Enter the *Password*, as provided by the service provider.

Registration status

See *Viewing registration status* on page 54.

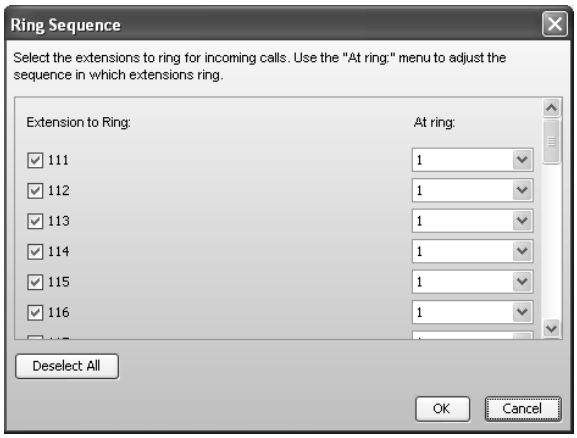
Call handling

The *Call Handling* area allows you to set up call handling for each VoIP number for Mode 1, Mode 2 and Holiday Mode. The VoIP number can ring selected local extensions in sequence, and then play an auto attendant or announcement, or go to a voice mailbox. Alternatively, it can immediately play an auto attendant or announcement, or go to a voice mailbox without ringing the extensions.

For example, the system can ring the receptionist when Mode 1 is active during the day. If the receptionist doesn't answer, it can start ringing other users as well. If there is still no answer, it can play an auto attendant. The auto attendant provides the dial-by-name directory, and allows the caller to dial an extension. A voice mailbox can immediately answer the VoIP number when Mode 2 is active at night or on weekends. Another auto attendant can immediately route the call to a remote extension during Holiday Mode.

1. Select the *Mode* tab. Choices are *Mode 1*, *Mode 2* and *Holiday Mode*.
2. Set up the ring sequence. The ring sequence determines when each local extension will start ringing. The extensions will not use their call cascades during the ring sequence. The system transfers the call to the first extension that is answered.

   a) Click **Edit**. The **Ring Sequence** window appears.



   b) Select the local extensions to ring. All extensions are selected by default.

   c) For each selected extension, set when the local extension will start ringing. All extensions ring after the first ring by default.

If you assign a different value to each extension, the extension with the lowest number of rings will ring first. The other extensions will start ringing as the number of rings goes up. For example, the system can ring the receptionist first, and then ring other users if the receptionist doesn't answer.

3. Set up call handling to route an unanswered call to an auto attendant or voicemail.

   a) Select the call handling action. Choices are:

      - *perform no action* — The system continues to ring the extensions. A generic auto attendant answers after 15 rings. It allows an authorized caller to make configuration changes, access voicemail or dial extensions.
      - *go to auto attendant* — Plays the selected auto attendant.
      - *go to voicemail* — Accesses the selected voice mailbox.
      - *play announcement* — Plays the selected announcement.

   b) If you selected *go to auto attendant*, select the auto attendant. You can assign the same auto attendant to multiple VoIP numbers, or a different auto attendant to each VoIP number.

   c) If you selected *go to voicemail* or *play announcement*, select the voice mailbox. You can select a local extension mailbox, remote extension mailbox, or general voice mailbox.

   d) Select when the system will perform call handling. Choices are:

      - *immediately* — The system will immediately perform call handling without using the ring sequence. The window will disable the **Edit** button and the other ring sequence controls.
      - *after 1 ring* to *after 9 rings* — The system will follow the ring sequence first, and then perform call handling if no one answers. The window enables the **Edit** button and the other ring sequence controls.

CALLER ID (OR CLID) BASED ROUTING

Incoming calls include caller ID information. The caller ID (referred to as CLID in some regions) includes the phone number and perhaps the name of the caller. The **Caller ID Based Routing** page allows you to set up call handling based on the caller ID information. The system will check the caller ID of each incoming call. If the caller ID matches a caller ID entry, the system will route the call accordingly.

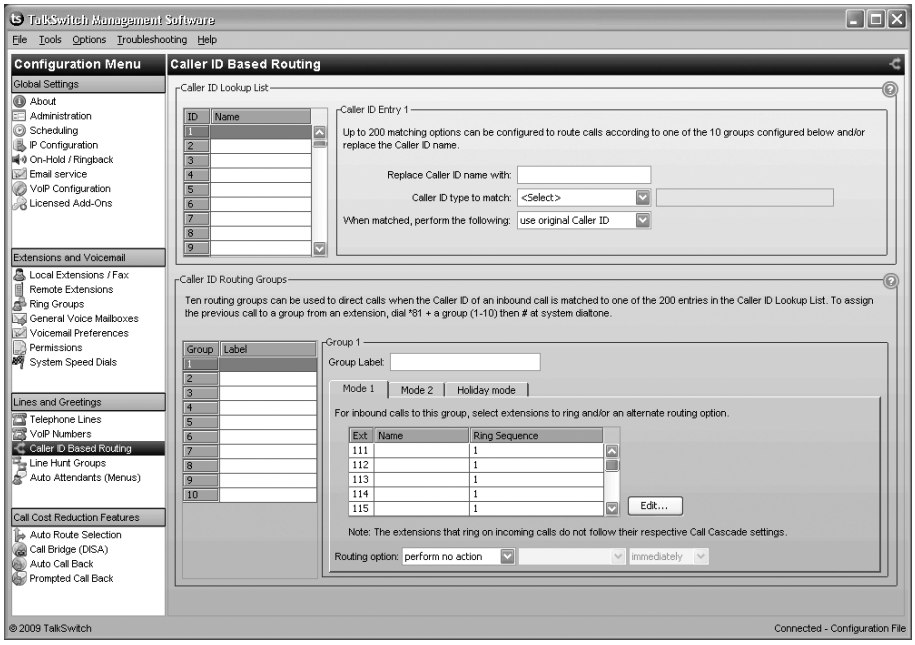
For example, if the call is from an important client, the call can be routed directly to the president's extension. If no caller ID is present, the call can be routed to voicemail.

You can define up to 200 caller ID entries. Each caller ID entry has an optional name, a phone number and a routing assignment. The routing assignment can use call handling set up for a group, or can display an alternate name on the user's extension, instead of the name from the caller ID.

If caller ID entries have phone numbers with overlapping digits, the system will use the best (longest) match to route the call. For example, the first entry routes calls from phone numbers that start with **5551**. The second entry routes calls from phone numbers that start with **555**. If the incoming call is from 555-1234, both entries match. However the system will use the first entry because it is a better match with a longer set of matching digits.

You can set up call handling for up to ten groups for Mode 1, Mode 2 and Holiday Mode. Call handling for a group is identical to call handling for telephone numbers, except you can set the ring pattern for each group. Depending on the caller ID, an incoming call can ring selected local extensions in sequence, and then play an auto attendant or announcement, or go to a voice mailbox. Alternatively, it can immediately play an auto attendant or announcement, or go to a voice mailbox, without ringing the extensions.

1. Select the **Caller ID Based Routing** page.


TalkSwitch Management Software
File Tools Options Troubleshooting Help
Configuration Menu
Caller ID Based Routing
Global Settings
About
Administration
Scheduling
IP Configuration
On-Hold / Ringback
Email service
VoIP Configuration
Licensed Add-Ons
Extensions and Voicemail
Local Extensions / Fax
Remote Extensions
Ring Groups
General Voice Mailboxes
Voicemail Preferences
Permissions
System Speed Dials
Lines and Greetings
Telephone Lines
VoIP Numbers
Caller ID Based Routing
Line Hunt Groups
Auto Attendants (Menus)
Call Cost Reduction Features
Auto Route Selection
Call Bridge (DISA)
Auto Call Back
Prompted Call Back
Caller ID Lookup List
ID
Name
Caller ID Entry 1
Up to 200 matching options can be configured to route calls according to one of the 10 groups configured below and/or replace the Caller ID name.
Replace Caller ID name with:
Caller ID type to match:
<Select>
When matched, perform the following:
use original Caller ID
Caller ID Routing Groups
Ten routing groups can be used to direct calls when the Caller ID of an inbound call is matched to one of the 200 entries in the Caller ID Lookup List. To assign the previous call to a group from an extension, dial *81 + a group (1-10) then # at system dialtone.
Group
Label
Group 1
Group Label:
Mode 1
Mode 2
Holiday mode
For inbound calls to this group, select extensions to ring and/or an alternate routing option.
Ext
Name
Ring Sequence
Edit...
Note: The extensions that ring on incoming calls do not follow their respective Call Cascade settings.
Routing option:
perform no action
immediately
© 2009 TalkSwitch
Connected - Configuration File

Note: *Caller ID* is referred to as *CLID* in some regions.

Caller ID lookup list or CLID matching list

The *Caller ID lookup list* area (or *CLID matching list* in some regions), allows you to define up to 200 caller ID entries. Each entry has an optional name, a phone number and a routing assignment. The routing assignment can use call handling set up for a group, or can display an alternate name on the user's extension, instead of the name from the caller ID.

1. Optionally, enter the name in the *Replace Caller ID name with* box. The extension will display this alternate name, instead of the name from the caller ID.
2. Select the *Caller ID type to match* or *CLID type to match*. Choices are:
   - *Private/blocked* — Defines an entry for incoming calls that have caller ID blocked for privacy reasons.
   - *Long distance/unknown* — Defines an entry for incoming calls that don't have caller ID.
   - *Phone number ends with* — Defines an entry for an incoming call from a phone number that ends with the specified digits. For example, you can enter an important client's phone number if you want to route their calls directly to the president's extension.

     All incoming calls from phone numbers with ending digits that match the specified digits will be routed the same way. For example, the specified digits are **5551212**. The system will use the same routing if the phone number is 555-1212, 613-555-1212 or 416-555-1212.
   - *Phone number starts with* — Defines an entry for an incoming call from a phone number that starts with the specified digits. For example, you can enter an area code if you want to route long distance calls to a particular extension for faster processing, or if you want to route telemarketers with 1-800 numbers directly to voicemail.

     All incoming calls from phone numbers with starting digits that match the specified digits will be routed the same way. For example, the specified digits are **1800**. The system will use the same routing if the phone number is 180-0121, 1-800-123-4567 or 1-800-555-1234.
3. If you selected *Phone number ends with* or *Phone number starts with*, enter the digits.
4. Select the routing in the *When matched, perform the following* list. Choices are:
   - *use original Caller ID* — The caller ID will not affect call routing.
   - *Replace Caller ID name* — The caller ID will not affect call routing, but the extension will display the alternate name, instead of the name from the caller ID.
   - *Route using Group [1–10]* — The caller ID is used to route the call according to the specified group. The extension will display the alternate name, instead of the name from the caller ID.

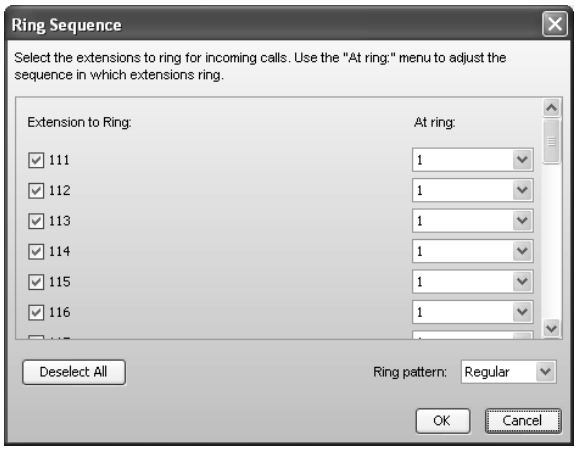
Caller ID or CLID Routing Groups

The *Caller ID Routing Groups* area (or *CLID Routing Groups* in some regions), allows you to set up call handling for up to ten groups for Mode 1, Mode 2 and Holiday Mode. Call handling for a group is identical to call handling for telephone numbers, except you can set the ring pattern for each group. Depending on the caller ID, an incoming call can ring selected local extensions in sequence, and then play an auto attendant or announcement, or go to a voice mailbox. Alternatively, it can immediately play an auto attendant or announcement, or go to a voice mailbox, without ringing the extensions.

For example, assume an important client is calling. The system can ring the president when Mode 1 is active during the day. If the president doesn't answer, the call can be routed to voicemail. A voice mailbox can immediately answer the call when Mode 2 is active at night or on weekends. An auto attendant can immediately route the call to the president's remote extension during Holiday Mode.

1. Enter the *Group label* to name the group.
2. Select the *Mode* tab. Choices are *Mode 1*, *Mode 2* and *Holiday Mode*.
3. Set up the ring sequence. The ring sequence determines when each local extension will start ringing. The extensions will not use their call cascades during the ring sequence. The system transfers the call to the first extension that is answered.

   a) Click **Edit**. The **Ring Sequence** window appears.



   b) Select the local extensions to ring. All extensions are selected by default.

   c) For each selected extension, set when the local extension will start ringing. All extensions ring after the first ring by default.

      If you assign a different value to each extension, the extension with the lowest number of rings will ring first. The other extensions will start ringing as the number of rings goes up. For example, the system can ring the receptionist first, and then ring other users if the receptionist doesn't answer.

   d) Select the *Ring pattern*. Choices are *Regular* and *Alternate 1* to *Alternate 5*. This allows you to set up distinctive ring based on the caller ID.

4. Set up call handling to route an unanswered call.

   a) Select the call handling action. Choices are:

      - *perform no action* — The system continues to ring the extensions. A generic auto attendant answers after 15 rings. It allows an authorized caller to make configuration changes, access voicemail or dial extensions.
      - *go to auto attendant* — Plays the selected auto attendant.
      - *play announcement* — Plays the selected announcement.
      - *go to voicemail* — Accesses the selected voice mailbox.

b) If you selected *go to auto attendant*, select the auto attendant. You can assign the same auto attendant to multiple groups, or a different auto attendant to each group.

c) If you selected *play announcement*, select the announcement.

d) If you selected *go to voicemail*, select the voice mailbox. You can select a local extension mailbox, remote extension mailbox, or general voice mailbox.

e) Select when the system will perform call handling. Choices are:

- *immediately* — The system will immediately perform call handling without using the ring sequence. The window will disable the **Edit** button and the other ring sequence controls.
- *after 1 ring* to *after 9 rings* — The system will follow the ring sequence first, and then perform call handling if no one answers. The window enables the **Edit** button and the other ring sequence controls.

Setting up caller ID routing with an extension

A user can set up caller ID routing with their extension, once the call is terminated. For example, a telemarketer can be added to a group that routes calls directly to an announcement.

1. To add the caller to a group, dial ***81 [group number 1–10] #**. The system adds the caller ID entry to the *Caller ID lookup list* (or *CLID matching list* in some regions). Future calls will be routed according to the group.
2. To display the alternate name instead of the caller ID name, dial ***810#**. The system adds the caller ID entry to the *Caller ID lookup list* (or *CLID matching list* in some regions). Future calls will display the name from the *Replace Caller ID name with* box.
3. To delete the caller ID entry from the *Caller ID lookup list* (or *CLID matching list* in some regions), dial ***81255#**. The system will no longer use the caller ID to route the call or display the alternate name.

LINE HUNT GROUPS

IMPORTANT: Ensure that hunt group 9 or 0 is assigned to the group of telephone lines or VoIP trunks used for calls to emergency services.

A line hunt group is a set of lines that are available for making an outbound call. It can use selected telephone lines, all VoIP lines associated with the TalkSwitch VoIP network, or all VoIP lines associated with a service provider VoIP network.

When placing an outbound call, the user first dials the hunt group number. The TalkSwitch system selects an available line from the group. However the user does not have to dial a hunt group before VoIP numbers 250–299: these automatically use the TalkSwitch VoIP network. Nor does the user have to dial a hunt group before remote extension numbers 210–249: these automatically use the hunt group configured for the remote extension.

You can set the hunt order for telephone lines, but the system automatically determines the hunt order for VoIP lines.

You can set up nine different hunt groups. If a VoIP-enabled TalkSwitch unit is present, hunt group 88 uses the TalkSwitch VoIP network by default, and the other hunt groups use telephone lines. You can modify these default settings as required. If you are using multiple service provider VoIP networks, set up a hunt group for each service provider.

A local extension can be restricted to a set of hunt groups, in order to reserve telephone lines for high-priority users, or to control access to VoIP networks. See *Hunt group permissions* on page 64.

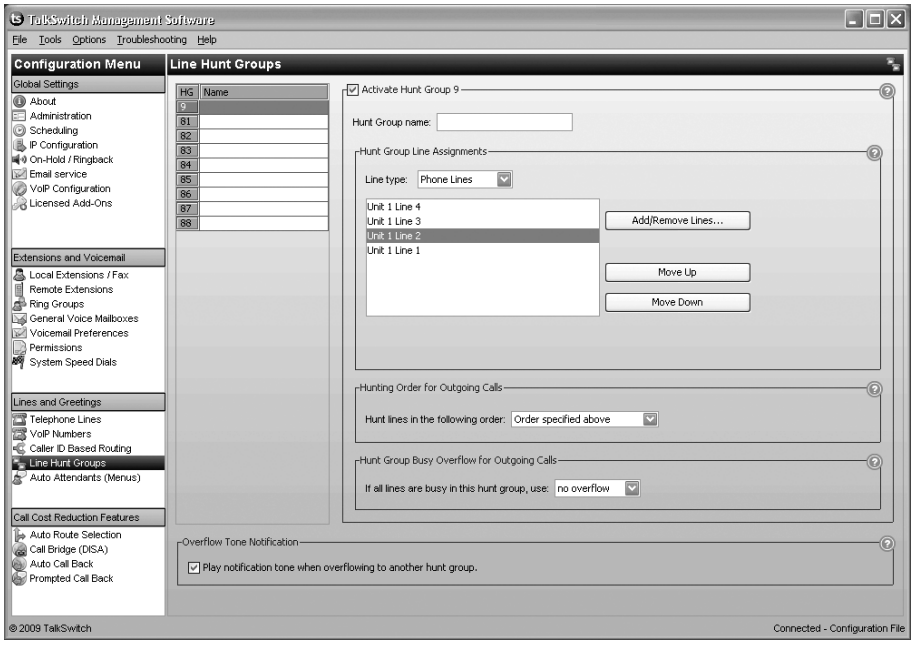
The TalkSwitch unit uses a hunt group when placing a call from a local extension, to a remote extension, or with the call bridge (DISA) feature. The hunt groups do not affect incoming calls.

The TalkSwitch system has the following hunt groups by default. If default settings are used, the system will first search the higher line numbers (e.g. line 4) for outgoing calls, since the lower line numbers are more heavily used for incoming calls. You can modify the default hunt groups as required.

Hunt Group 9 (0 in some regions)	Selects line 4, line 3, line 2, then line 1.
Hunt Group 81	Selects line 1.
Hunt Group 82	Selects line 2.
Hunt Group 83	Selects line 3.
Hunt Group 84	Selects line 4.
Hunt Group 85–87	No lines are assigned.
Hunt Group 88	If a VoIP-enabled TalkSwitch unit is present, selects an available VoIP line from those associated with the TalkSwitch VoIP network. If no VoIP-enabled TalkSwitch unit is present, no lines are assigned.

The **Line Hunt Groups** page allows you to set up the name, line type, set of telephone lines, hunting order, overflow hunt group and overflow notification for each hunt group.

1. Select the **Line Hunt Groups** page.



Note: Depending on the region, an operator may be dialed using 9 or 0.

Activate hunt group

The *Activate Hunt Group* area allows you to activate and name a hunt group.

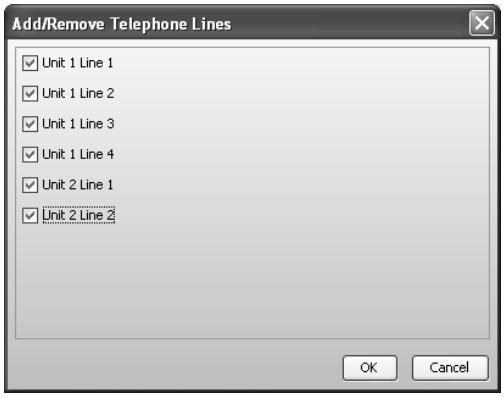
1. Select a hunt group.
2. If necessary, select the *Activate Hunt Group* checkbox. All hunt groups are active by default.
3. Enter a name in the *Hunt group name* box. The name will identify the hunt group elsewhere in the TalkSwitch management software.

Hunt group line assignments

The *Hunt Group Line Assignments* area allows you to set up a hunt group for telephone lines or VoIP lines.

1. Set the *Line type*. Choices are:
   - *Phone lines* — Allows you to select telephone lines.
   - *TalkSwitch VoIP* — Uses the TalkSwitch VoIP network lines in the hunt group.
   - *SPn VoIP Service* — Uses the service provider VoIP lines in the hunt group.
2. If you set *Line type* to *Phone lines*:

a) Click **Add/Remove Lines**. The **Add/Remove Telephone Lines** window appears.



b) Select the telephone lines for the hunt group, and then click **OK**.

c) Change the order of the lines by selecting a line, and then clicking **Move Up** and **Move Down**.

If you set *Line type* to *TalkSwitch VoIP* or *SPn VoIP Service*, VoIP lines are selected automatically.

Hunting order for outgoing calls

The *Hunting order for outgoing calls* area allows you to specify how the system will hunt for an available line within the hunt group.

1. Set the *Hunt lines in the following order* list. Choices are:

   - *Order specified above* — The system will start with the first line in the hunt group.
   - *Search lines on same unit first* — The system will hunt for lines on the TalkSwitch unit that the extension is connected to, before hunting on the other units. This minimizes network traffic between TalkSwitch units.
   - *Uniform distribution (Circular)* — The system will start with the line after the last hunted line in the hunt group. This way telephone lines are used more evenly for outbound calls.

     For example, the hunt group has lines 1, 2, 3 and 4. The last call with the hunt group used line 2. The system will first check line 3 for availability.

     The circular hunt mode can reduce costs if the line toll increases after utilization exceeds a threshold. For example, assume each line costs 5¢ per minute for the first 100 minutes, and then costs 10¢ per minute. The circular hunt mode makes it more likely users will finish the first 100 minutes on each line before the thresholds are exceeded and higher charges are incurred.

Hunt group busy overflow for outgoing calls

The *Hunt group busy overflow for outgoing calls* area allows you to select an overflow hunt group. If the user dials a hunt group, but there are no lines available, the system will play a warning tone while hunting for a free line in the overflow hunt group. The warning tone indicates the call may be more expensive or of different quality than expected. The user can then remain on the line, or hang up and wait for a line to become available in the original hunt group.

If the dialed hunt group and the overflow hunt group are both busy, the user will hear the busy tone.

A hunt group that contains telephone lines can have an overflow hunt group that contains telephone lines or VoIP lines. Similarly, a hunt group that contains VoIP lines can have an overflow hunt group that contains telephone lines or VoIP lines.

The overflow hunt group must contain different lines than the original hunt group, but these lines must support the same dialed number structure.

For example, assume the original hunt group contains telephone lines and the overflow hunt group contains VoIP lines. If the lines within the original hunt group require you to dial **1** before the area code and phone number (e.g. 1 613 555 1212), the lines within the overflow hunt group cannot require you to omit the **1** (e.g. 613 555 1212).

1. Select the overflow hunt group in the *If all lines are busy in this hunt group* list. Choices are *no overflow* and the activated hunt groups.

Overflow tone notification

The *Overflow tone notification* area allows you to disable the warning tone that sounds when an overflow hunt group is being used. The warning tone sounds by default.

1. To disable the warning tone, clear the *Play notification tone when overflowing to another hunt group* checkbox.

AUTO ATTENDANTS (MENUS)

An auto attendant can answer a telephone line or VoIP number, and can be included in the call cascade of a local extension, remote extension or ring group.

An auto attendant can answer a call if the receptionist is away or if you don't have a receptionist. Each auto attendant has a message and up to six options. The message tells the caller what the options are. You can load a professionally pre-recorded message, or can record a message using a handset. The caller selects an option by dialing **0** (**9** in some regions), **1**, **2**, **3**, **4** or **5**. The auto attendant then performs the action programmed for the option. The auto attendant can:

- Transfer the call to a local extension, remote extension or ring group. The call then follows the extension's call cascade.
- Transfer the call to the call queue of a ring group. The call is placed on hold. The TalkSwitch system will ring the next available local extension in the ring group.
- Transfer the call to a voice mailbox, allowing the caller to leave a message. The call can be transferred to a local extension, remote extension or general voice mailbox. Pressing ***** during the greeting returns to the auto attendant. If the **0** option is programmed (**9** in some regions), pressing **0**during the greeting (**9** in some regions) can route the call to an extension, voice mailbox, announcement or auto attendant. An authorized caller can retrieve messages and perform other voicemail activities by pressing **8** during the greeting. If no digit is pressed during the greeting, the system hangs up after the caller leaves a message.
- Play an announcement with directions, business hours, etc. The announcement can have the **0** option programmed (**9** in some regions) to route the call to an extension, voice mailbox, announcement or auto attendant. The system hangs up after the announcement if no digit is dialed. Pressing ***** returns to the auto attendant.
- Access the dial-by-name directory so the caller can find a user's extension number. The dial-by-name directory prompts the caller to enter the first three letters of the user's last name. Pressing ***** returns to the auto attendant. See *Setting up the dial-by-name directory* on page 158.
- Route the call to another auto attendant, which allows actions to be nested into a powerful call routing system. For example, the main auto attendant can say *"Press one for English. Oprima dos para Español."* Option 1 goes to the English auto attendant and option 2 goes to the Spanish auto attendant.

In addition to the six auto attendant options, the caller can:

- Reach an extension by dialing an extension number.
- Access voicemail by pressing ****** then the mailbox number.
- Access call back by dialing **6**.
- Access call bridge/DISA by dialing a hunt group number.
- Enter command mode by pressing **#**.

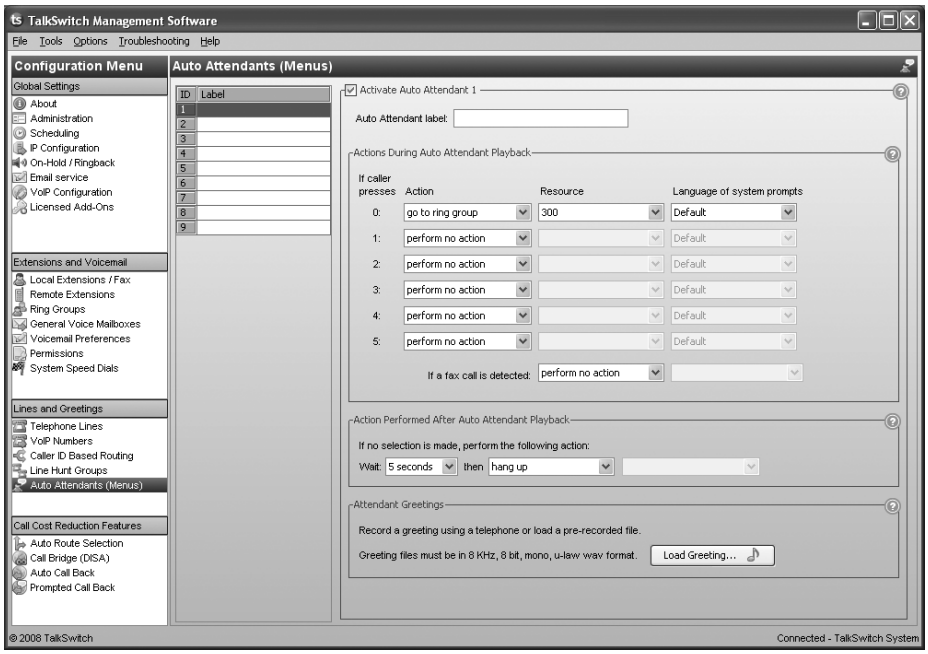
The caller is able to dial an extension, even if the first number of the extension is the same as an auto attendant option. For example, the caller can dial **111** even though the first "1" is the same as auto attendant option 1. This is because the TalkSwitch system waits after the caller dials a digit, before following the action for that digit. The default wait time is 1.5 seconds, and is set with the *Single digit fall through time* list in the **Auto Attendants** window.

The TalkSwitch system automatically copies the auto attendants to each TalkSwitch unit on the network. This reduces network traffic and allows the system to continue functioning even if a unit loses power or is disconnected from the LAN.

The recording time for internal music on hold, voicemail, and the auto attendants is shared on the TalkSwitch unit. You can expand the built-in memory using memory cards.

The **Auto Attendants (Menus)** page allows you to set up the nine different auto attendants for the TalkSwitch system.

1. Select the **Auto Attendant (Menus)** page.



Note: In the *Actions During Auto Attendant Playback* section, depending on the region, callers can press either 0 or 9.

Activate auto attendant

The *Activate Auto Attendant* area allows you to activate an auto attendant. All auto attendants are activated by default.

1. Select an auto attendant.
2. If necessary, select the *Activate Auto Attendant* checkbox.
3. Enter the *Auto attendant label*. The name will identify the auto attendant elsewhere in the TalkSwitch management software.

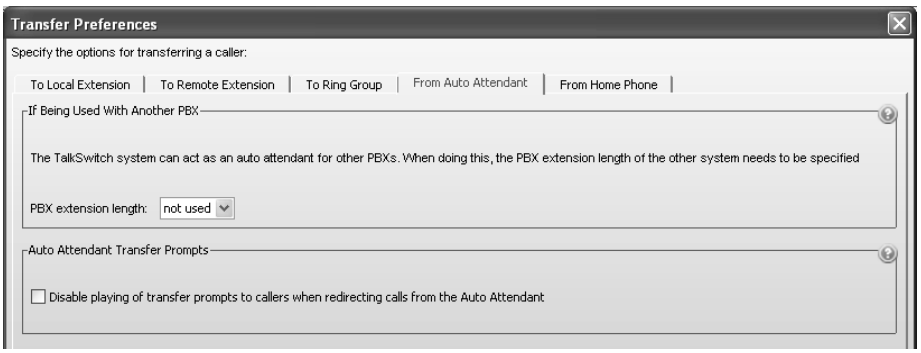
Actions during auto attendant playback

The *Actions During Auto Attendant Playback* area allows you to configure the auto attendant with up to six options, associated with the caller pressing **0** (**9** in some regions), **1**, **2**, **3**, **4** or **5**.

1. Select the *Action*. Choices are:
   - *go to voicemail* — Accesses the selected voice mailbox.
   - *go to local extension* — Transfers the call to the selected local extension.
   - *go to remote extension* — Transfers the call to the selected remote extension.

- *go to ring group* — Transfers the call to the selected ring group.
- *play announcement* — Plays the selected announcement.
- *go to auto attendant* — Routes the call to the selected auto attendant.
- *queue at ring group* — Transfers the call to the call queue of the selected ring group.
- *lookup name directory* — Accesses the dial-by-name directory. See *Setting up the dial-by-name directory* on page 158.
- *perform no action* — The option is unused.
- *go to VoIP location* — Routes the call to the selected VoIP number within the TalkSwitch VoIP network.

2. Select the *Resource*. Depending on the action selected in Step 1, resources are voice mailboxes, extensions, announcements, auto attendants, or VoIP numbers within the TalkSwitch VoIP network.
3. Select the language in the *Language of system prompts* list. If the caller selects this option, they will hear all subsequent prompts in the selected language.
4. Select the action *If a fax call is detected*. The TalkSwitch system can automatically detect a fax machine that plays a CNG tone. Choices are:
   - *go to local extension*
   - *go to remote extension*
   - *perform no action*
   - *hang up*
5. Depending on the action selected in Step 4, select the extension the fax machine is connected to.
6. If the caller dials an option that causes the auto attendant to transfer their call, the system will play the "*One moment please*" prompt by default. You can disable the prompt. See *From auto attendant tab* on page 186.


Transfer Preferences
Specify the options for transferring a caller:
To Local Extension
To Remote Extension
To Ring Group
From Auto Attendant
From Home Phone
If Being Used With Another PBX
The TalkSwitch system can act as an auto attendant for other PBXs. When doing this, the PBX extension length of the other system needs to be specified
PBX extension length:
not used
Auto Attendant Transfer Prompts
Disable playing of transfer prompts to callers when redirecting calls from the Auto Attendant

   a) Choose **Options > Transfer Preferences**. The **Transfer Preferences** window appears.
   b) Select the *From Auto Attendant* tab.
   c) To disable the prompt, select the *Disable playing of transfer prompts* checkbox.

Action performed after auto attendant playback

The *Action Performed After Auto Attendant Playback* area allows you to set up how the system will react if the user makes no selection. This can occur if the caller does not understand the prompts, does not have tone dialing on their phone, or prefers to speak with a person.

1. Select the time limit, ranging from *0 seconds* to *30* seconds.
2. Select the action. Choices include:
   - *go to voicemail*
   - *go to local extension*
   - *go to remote extension*
   - *go to ring group*
   - *play announcement*
   - *go to auto attendant*
   - *queue at ring group*
   - *lookup name directory* — See *Setting up the dial-by-name directory* on page 158.
   - *hang up*
3. Depending on the action selected in Step 2, select the voice mailbox, extension, announcement, or auto attendant.

Working with auto attendant messages

You can record a new message, load a professionally recorded message, or erase a message from the TalkSwitch system.

If you don't set up an auto attendant message, the TalkSwitch system will answer without playing a message. The caller will hear silence and might assume the call did not get connected.

Recording a new message

You can record a new message using a local extension or remote phone.

1. Pick up a local extension, or dial into the TalkSwitch system from a remote phone. If you pick up a local extension, you will hear the dial tone. If you dial in from a remote phone, the auto attendant will answer.
2. Enter command mode by either pressing **#** on an analog extension phone, or ***55#** on a TalkSwitch IP phone (note: other brands may use ***55 Send** or ***55 Dial**).
3. Enter the system password, followed by **#**.
4. Dial **4 [auto attendant number 1 - 9] #** to record the message. For example, dial **41#** to record the message for auto attendant 1.
5. Press **#** when you have completed saying the message.
6. Dial **5 [auto attendant number 1 - 9] #** to listen to the message. For example, dial **51#** to listen to the message for auto attendant 1.

7. Repeat Steps 4 to 6 to re-record the message, or hang up to keep the message.

Loading a message

The *Attendant Greetings* area allows you to load the message for the auto attendant.

You can load an 8 kHz, 8 bit, Mono, u-Law .wav file that contains the auto attendant message. If using a professionally recorded message, ensure it is in this format.

1. Select an auto attendant.
2. Click **Load Greeting**. The **Select Wav File** window appears.

3. Click **Browse** to select the .wav file, and then click **Select Wav File**.

Erasing a message

You can erase a recorded or loaded message using a local extension or remote phone. Erase unused auto attendant messages to free up space for voicemail.

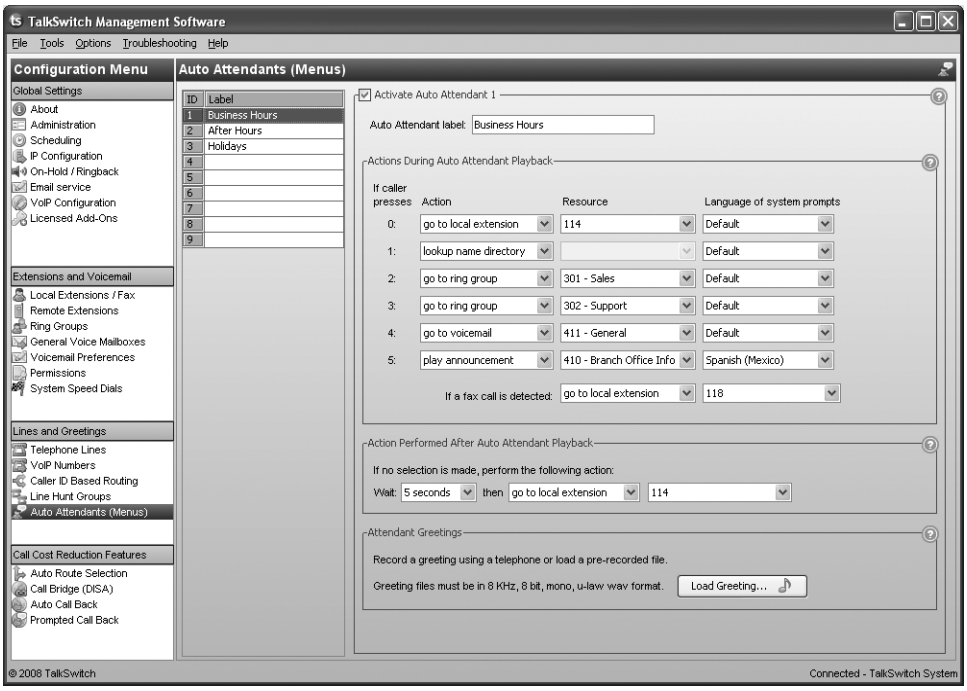
1. Pick up a local extension, or dial into the TalkSwitch system from a remote phone. If you pick up a local extension, you will hear the dial tone. If you dial in from a remote phone, the auto attendant will answer.
2. Enter command mode by either pressing **#** on an analog extension phone, or ***55#** on a TalkSwitch IP phone (note: other brands may use ***55 Send** or ***55 Dial**).
3. Enter the system password followed by **#**.
4. Dial **04 [auto attendant number 1 - 9] #** to erase the message. For example, dial **041 #** to erase the message for auto attendant 1.

Example auto attendant

The following auto attendant answers incoming calls, and plays the following recorded message:

"Welcome to ABC Company. If you know your party's three-digit extension, you may dial it now. To reach the receptionist, dial 0 (9 in some regions) *or stay on the line. Dial 1 to find your party's extension in our dial-by-name directory. Dial 2 to hear our business hours and directions to our location. Dial 3 to talk to the next available customer support representative, or dial 4 to leave them a message."*

If the caller is a fax machine that plays the CNG tone, the TalkSwitch system will route the call to the fax machine at extension 118. If no selection is made, the call is routed to the receptionist at extension 114.



Note: In the *Actions During Auto Attendant Playback* section, depending on the region, callers can press either 0 or 9.

Setting up the dial-by-name directory

The dial-by-name directory allows a caller to find a user's extension number, and connect to their local extension or remote extension. This way the caller can reach their party without speaking to the receptionist.

When prompted by the auto attendant, the caller selects the dial-by-name directory, and then dials the first three letters of the user's last name. Alternatively, the caller can dial only one or two letters. If a matching entry is found, the system will play the user's name and extension number. The caller can then dial **1** to connect to the user's extension. If there are multiple matches, the caller can dial **2** to hear the next matching name and extension number.

To make use of the dial-by-name directory:

1. Set up the local extensions (see *Activate extension* on page 58) and remote extensions (see *Activate extension* on page 117). The *Last name* from each extension will be used in the directory.

2. Have each user set up their voice mailbox by dialing ****#** and then following the prompts. They will record their names for the dial-by-name directory. Alternatively, you can set up their voice mailboxes by dialing ****** + voice mailbox number + **#** and then following the prompts.

The name must be recorded for the entry to be available.

3. Configure the telephone line or VoIP number to be answered by an auto attendant. See *Call handling* on page 136.

4. Configure the auto attendant with an action to access the lookup name directory. See *Actions during auto attendant playback* on page 154.

5. Record an announcement for the auto attendant that tells the caller how to access the dial-by-name directory. See *Working with auto attendant messages* on page 156.

6. To make the dial-by-name directory available to users as well as callers, configure the Dial 0 Routing feature (Dial 9 in some regions), to connect with the lookup name directory. See *Dial 9 or 0 routing* on page 28.

You can delete the names recorded for the dial-by-name directory by resetting the voice mailbox. See *Reset mailboxes* on page 125.

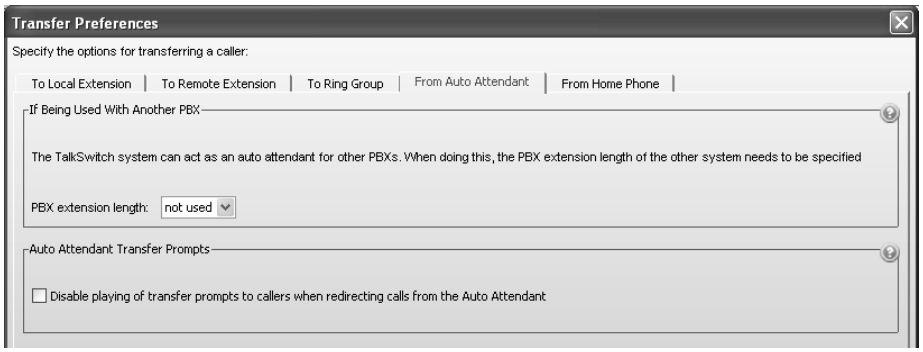
Setting the PBX extension length

You can configure a TalkSwitch unit with an auto attendant, and then connect the unit to another PBX. This way the TalkSwitch unit can route callers as requested. See *Connecting a TalkSwitch unit to a PBX* on page 21.

If you will connect the TalkSwitch unit to another PBX, set the extension length used by the other PBX. See *From auto attendant tab* on page 186.

1. Choose **Options > Transfer Preferences**. The **Transfer Preferences** window appears.

2. Select the *From Auto Attendant* tab.


Transfer Preferences
Specify the options for transferring a caller:
To Local Extension
To Remote Extension
To Ring Group
From Auto Attendant
From Home Phone
If Being Used With Another PBX
The TalkSwitch system can act as an auto attendant for other PBXs. When doing this, the PBX extension length of the other system needs to be specified
PBX extension length:
not used
Auto Attendant Transfer Prompts
Disable playing of transfer prompts to callers when redirecting calls from the Auto Attendant

3. Set the *PBX extension length* to the number of digits in an extension number on the other PBX.

AUTO ROUTE SELECTION

Introduction

Enable password protection when ARS or toll restriction is used. Change the system password frequently to prevent unauthorized users from making calls or changing the configuration. See *Administration* on page 27.

About automatic route selection

Automatic route selection (ARS) routes outgoing calls to a certain hunt group depending on the leading digits. It can also prefix phone numbers with certain carrier codes depending on the leading digits and time of day. This way calls can use specific telephone lines, alternative carriers, or VoIP lines, for least-cost routing. For example, ARS can route international calls to a VoIP service provider; national calls to an alternative carrier offering a discounted calling plan; and calls to cellular phones on a telephone line that is connected to a fixed wireless terminal for cellular-to-cellular calling.

If the user dials a hunt group number, ARS can override the selection and use a different hunt group depending on the leading digits.

About toll restriction

Toll restriction blocks outgoing calls based on the leading digits. For example, toll restriction can block calls to inappropriate or premium rate services (e.g. 1-900 numbers).

About leading digits

The leading digits are the first numbers dialed when placing a telephone call. The leading digits could be an international access code, country code, national code, STD code, area code, leading digits of premium rate service codes, or phone number. For example, “1900” are the leading digits of 1-900-555-1234.

The leading digits used in each country are subject to the national numbering plan. Your telephone directory will provide information on the numbers used.

Calls with leading digits that do not match the entries in the ARS table will be routed to the hunt group originally dialed by the caller.

ARS and toll restriction act on the longest leading digits entry matching the dialed number. For example, an entry in the ARS table blocks numbers with leading digit “1” to prevent long-distance calls. However a second entry routes numbers with leading digits “1800” through a hunt group because they are toll-free. Because the second entry (1800) is longer than the first entry (1), the second entry has precedence. Therefore a call with leading digits “1800” is routed, even though leading digit “1” would otherwise cause it to be blocked. This technique can reduce the number of entries required in the ARS table; and can be used to restrict calls to all but specific area codes or countries.

- Leading digits can be 1–11 numbers in length.
- Leading digits can include numbers 0–9.
- The leading digits exclude the hunt group number dialed by the user.

About carrier codes

A carrier code is prefixed to the phone number dialed by the user. It tells the telephone company to route the call to an alternative carrier. For example, the carrier code could be a calling card number and PIN number. ARS determines the carrier code based on the leading digits and the time of day.

- Carrier codes can be up to 1–24 characters in length.
- Carrier codes can include numbers **0**–**9**, "*****" (asterisk), "**#**" (pound), "**,**" (comma for 2-second pause), and "**w**" (for wait-for-dial-tone). A 2-second pause is automatically added after dialing a carrier code.

An entry in the ARS table can have one or two carrier codes. *Carrier code 1* is either used all day, or from *Start time 1* until *Start time 2*. *Carrier code 2* is used from *Start time 2* until *Start time 1*.

Use of Carrier Selection Prefix and Carrier Codes may require a subscription to a carrier's discount calling plan. This service may not be available in some countries and on some telephone companies' telephone lines. Carrier Selection Prefix may not be required if a "Pre-Selection" service is provided by the telephone company.

ARS and direct line access

When Direct Line Access is enabled on a local extension, the user does not have to dial a hunt group number. Calls placed from that local extension are automatically routed to a designated hunt group. If ARS is enabled on the designated hunt group, the calls are subject to ARS and toll restriction.

ARS and the 3-Way Calling/Conference service

The use of the telephone company's 3-Way Calling/Conference service is not recommended on telephone lines when ARS or toll restriction are used. The system's ARS and toll restriction features do not have any control over the routing of calls through these telephone company services.

ARS and emergency service numbers

The telephone numbers in the *Emergency Service Numbers* area will not be blocked by toll restriction nor redirected by ARS. Calls to these telephone numbers will be routed through hunt group 9 (0 in some regions).

Enter any other local emergency service numbers into the ARS table to ensure calls are routed to a hunt group with telephone lines that provide access to emergency services.

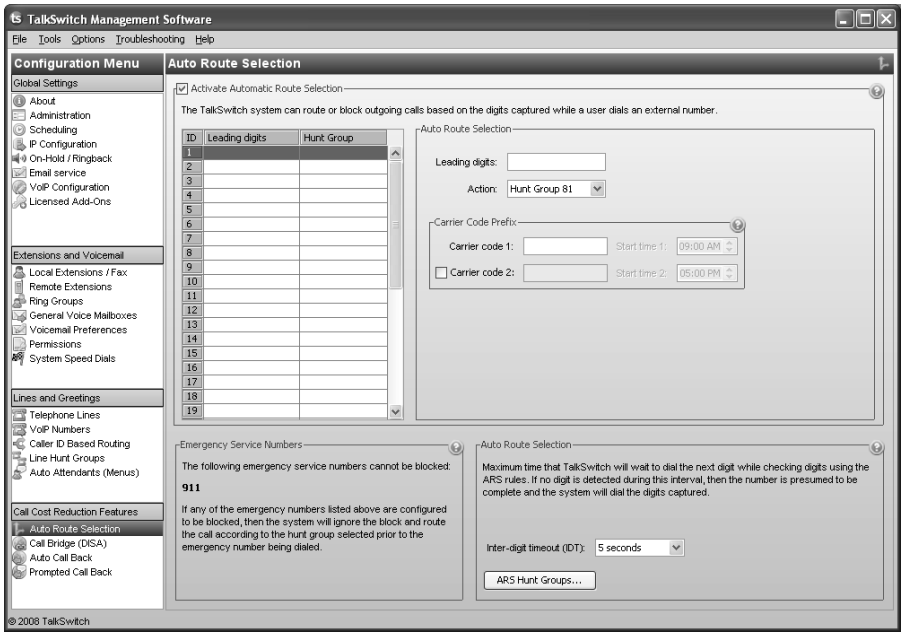
Planning automatic route selection and toll restriction

1. Identify the leading digits of calls to be blocked by toll restriction.
2. Identify the leading digits of calls to be routed by ARS.
3. Identify which telephone lines and VoIP lines are connected to different service providers.
4. Assign telephone lines and VoIP lines to different hunt groups, in order to provide separate access to each service provider.
5. Identify carrier codes, which leading digits they work with, and when they will be used.
6. Provide local extensions with access to specific hunt groups. This controls how local extensions have calls routed by ARS or blocked by toll restriction.

Setting up automatic route selection and toll restriction

The **Auto Route Selection** page allows you to set up automatic route selection (ARS) and toll restriction.

1. Configure the system password and hunt groups before setting up ARS and toll restriction.
2. Select the **Auto Route Selection** page.


TalkSwitch Management Software
File Tools Options Troubleshooting Help
Configuration Menu
Global Settings
About
Administration
Scheduling
IP Configuration
On-Hold / Ringback
Email service
VoIP Configuration
Licensed Add-Ons
Extensions and Voicemail
Local Extensions / Fax
Remote Extensions
Ring Groups
General Voice Mailboxes
Voicemail Preferences
Permissions
System Speed Dials
Lines and Greetings
Telephone Lines
VoIP Numbers
Caller ID Based Routing
Line Hunt Groups
Auto Attendants (Menus)
Call Cost Reduction Features
Auto Route Selection
Call Bridge (DISA)
Auto Call Back
Prompted Call Back
Auto Route Selection
Activate Automatic Route Selection
The TalkSwitch system can route or block outgoing calls based on the digits captured while a user dials an external number.
ID
Leading digits
Hunt Group
Auto Route Selection
Leading digits:
Action:
Hunt Group 81
Carrier Code Prefix
Carrier code 1:
Start time 1:
09:00 AM
Carrier code 2:
Start time 2:
05:00 PM
Emergency Service Numbers
The following emergency service numbers cannot be blocked:
911
If any of the emergency numbers listed above are configured to be blocked, then the system will ignore the block and route the call according to the hunt group selected prior to the emergency number being dialed.
Auto Route Selection
Maximum time that TalkSwitch will wait to dial the next digit while checking digits using the ARS rules. If no digit is detected during this interval, then the number is presumed to be complete and the system will dial the digits captured.
Inter-digit timeout (IDT):
5 seconds
ARS Hunt Groups...
© 2008 TalkSwitch

Activate automatic route selection

The *Activate Automatic Route Selection* area allows you to activate automatic route selection, and list the leading digits of calls to be re-directed or blocked.

1. Select the *Activate Automatic Route Selection* checkbox. When activated, the dialled digits of all outgoing calls will be buffered and stored before being re-transmitted to the telephone line after detection of dial tone. This feature is useful in cases where extension users forget to wait for the telephone line dial tone (secondary dial tone) when making outgoing calls.
2. Select a free table entry to enter new leading digits, or edit an existing entry.
3. Enter the leading digits in the *Leading digits* box. Omit the hunt group number from the leading digits. The system will only block or redirect calls where there is a matching leading digit match and action; calls with leading digits not entered in the table will proceed without being re-directed or blocked.
4. Select the *Action*. Choices are:
   - *Hunt group (n)* — Performs ARS by routing the call to the hunt group.
   - *Block calls* — Performs toll restriction by blocking the call.

Notes: The system selects the required action by matching the longest leading digits in the ARS table to the digits dialled by the caller. Typical leading digits to enter in the table are international and trunk call prefixes, area codes, premium service codes, and mobile service provider codes.

Carrier code prefixes

The *Carrier Code Prefix* area allows you to set up carrier codes the system will automatically use when the leading digits are dialed. A carrier code is prefixed to the phone number dialed by the user. It tells the telephone line service provider to route the call to an alternative carrier. For example, the carrier code could be a calling card number and PIN number. ARS determines the carrier code based on the leading digits and the time of day.

1. If the route requires one carrier code for the whole day:
    a) Enter the carrier code in *Carrier code 1*.
    b) Clear the *Carrier code 2* checkbox.
2. If the route requires two carrier codes, one for each part of the day:
    a) Enter the first carrier code in *Carrier code 1*.
    b) Enter *Start time 1*.
    c) Select the *Carrier code 2* checkbox.
    d) Enter the second carrier code in *Carrier code 2*.
    e) Enter *Start time 2*.

Emergency service numbers

The *Emergency Service Numbers* area lists the emergency service numbers that cannot be blocked by toll restriction.

Auto route selection

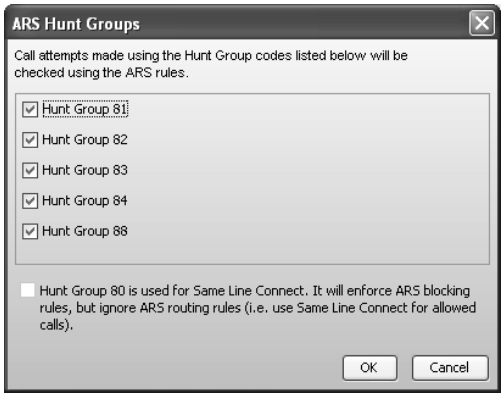
The *Auto Route Selection* area allows you to set the inter-digit timeout (IDT), and to select the hunt groups that are subject to ARS.

1. Select the *Inter-digit timeout (IDT)*. The default value is *5 seconds*. Choices range from *3 seconds* to *18 seconds*.

    ARS and toll restriction are implemented on each call once the longest matching entry is found in the ARS table or after the IDT expires. If the user has dialed a number and it's a possible match for an entry in the ARS table, the system will pause until the IDT expires. To complete a dialed number before the IDT expires, the user can press **#**.

    For example, the ARS table has 12345. The user dials hunt group 88 to use the TalkSwitch VoIP network, and dials VoIP number 1234. This is a possible match for entry 12345, so system will pause until the IDT expires. The user can press **#** to complete the call.
2. Select the hunt groups that are subject to ARS and toll restriction.

a) Click **ARS Hunt Groups**. The **ARS Hunt Groups** window appears.


ARS Hunt Groups
Call attempts made using the Hunt Group codes listed below will be checked using the ARS rules.
Hunt Group 81
Hunt Group 82
Hunt Group 83
Hunt Group 84
Hunt Group 88
Hunt Group 80 is used for Same Line Connect. It will enforce ARS blocking rules, but ignore ARS routing rules (i.e. use Same Line Connect for allowed calls).
OK
Cancel

b) Select the *Hunt group* checkboxes. The selected hunt groups will be subject to ARS and toll restriction.

c) If *3-Way Calling/Conference* is selected in the **Telephone Lines** page, the *Hunt group 80 is used for same line connect* checkbox is enabled. If the checkbox is selected, and a remote user enters hunt group 80 to activate call bridge, the same telephone line will be used to place the call (instead of a telephone line chosen by ARS). However toll restriction will still be used.

Restricting local extension access to specific hunt groups

By default, all local extensions have access to all hunt groups. As part of an ARS and toll restriction scheme you can restrict the hunt groups that a local extension can use.

1. Remove the local extension's access to the hunt group. See *Hunt group permissions* on page 64.

NOTE: If the local extension does not have access to a hunt group, but the user dials leading digits that select a hunt group, the hunt group is used anyway.

Allowing a local extension to bypass ARS and toll restriction

You can allow a local extension to select a hunt group that bypasses ARS and toll restriction. The user will be able to dial any phone number, and no carrier codes will be prefixed to the phone number.

1. Set up a hunt group with telephone lines or VoIP lines. See *Line Hunt Groups* on page 149.
2. Provide the local extension with access to the hunt group. See *Hunt group permissions* on page 64.
3. Select the **Auto Route Selection** page.
4. Click **ARS Hunt Groups**. The **ARS Hunt Groups** window appears.
5. Clear the associated *Hunt group* checkbox. The hunt group will not be subject to ARS and toll restriction.

Regulatory advisory notice

Call Redirection & Service Provider Billing Advisory

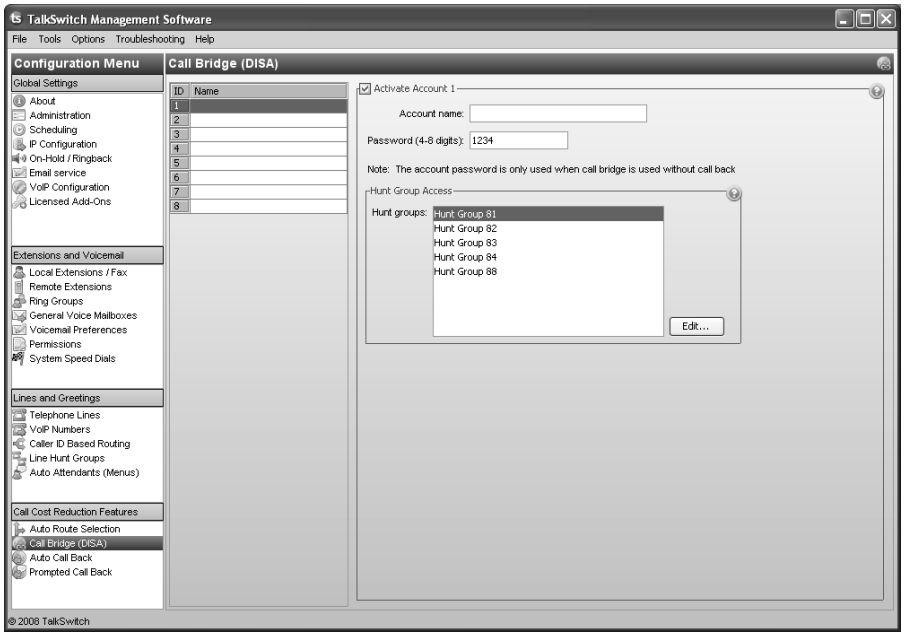
Use of the automatic route selection, toll restriction and call detail recording features does not imply any guarantee whatsoever by regulatory authorities, your telephone service provider(s), TalkSwitch or its distributors and resellers, with regard to the accuracy of these features and that the use of such a features may not be considered by a telephone company in any disputes which may arise regarding the accuracy of any subscriber's telephone account.

CALL BRIDGE (DISA)

Call bridge lets a travelling user place a call through the TalkSwitch system using your company's long-distance savings plan. This minimizes long-distance and hotel telephone costs. You can set up eight call bridge accounts for the system.

The **Call Bridge (DISA)** page allows you to set up the name, password and hunt groups for up to eight call bridge accounts.

1. Select the **Call Bridge (DISA)** page.



Using call bridge

1. Call the TalkSwitch system, and let the auto attendant answer.
2. Dial a hunt group to get an outside line.
3. Enter the call bridge password + **#**.
4. Dial the phone number.
5. To complete the call, you can:
   - Press **#*** to return to the auto attendant.
   - Press **##** to call another outside number using the same hunt group.

You may incur long-distance and hotel telephone charges each time you call the TalkSwitch system from a remote location.

Activate account

The *Activate Account* area allows you to activate and name a call bridge account.

1. Select an available call bridge account.
2. Select the *Activate Account* checkbox.

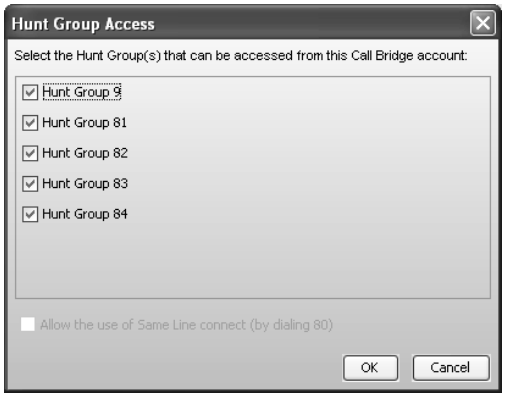
3. Enter an *Account name*. This is typically the name of the user, or the name of a department whose members will share the call bridge account.

4. To prevent unauthorized use of the TalkSwitch system, enter a numeric *Password*. The user will be required to enter this password before dialing the phone number.

Hunt group access

The *Hunt Group Access* area allows you to select the hunt groups that can be used with the call bridge account.

1. Click **Edit**. The **Hunt Group Access** window appears.


Hunt Group Access
Select the Hunt Group(s) that can be accessed from this Call Bridge account:
Hunt Group 9
Hunt Group 81
Hunt Group 82
Hunt Group 83
Hunt Group 84
Allow the use of Same Line connect (by dialing 80)
OK
Cancel

Note: Depending on the region, callers can access either Hunt Group 9 or Hunt Group 0.

2. Select the hunt groups that can be used with the call bridge account.

3. If *3-Way Calling/Conference* is selected in the **Telephone Lines** page, the **Hunt Group Access** window enables the *Allow the use of same line connect (by dialing 80)* checkbox. Select the checkbox to enable hunt group 80.

   If the user activates call bridge by dialing hunt group 80, the TalkSwitch system will use the same telephone line to call the third party. It will have the telephone company put the user on hold. It will then dial the third party's phone number, using the same telephone line. Once the third party answers, the TalkSwitch system will connect the user and the third party.

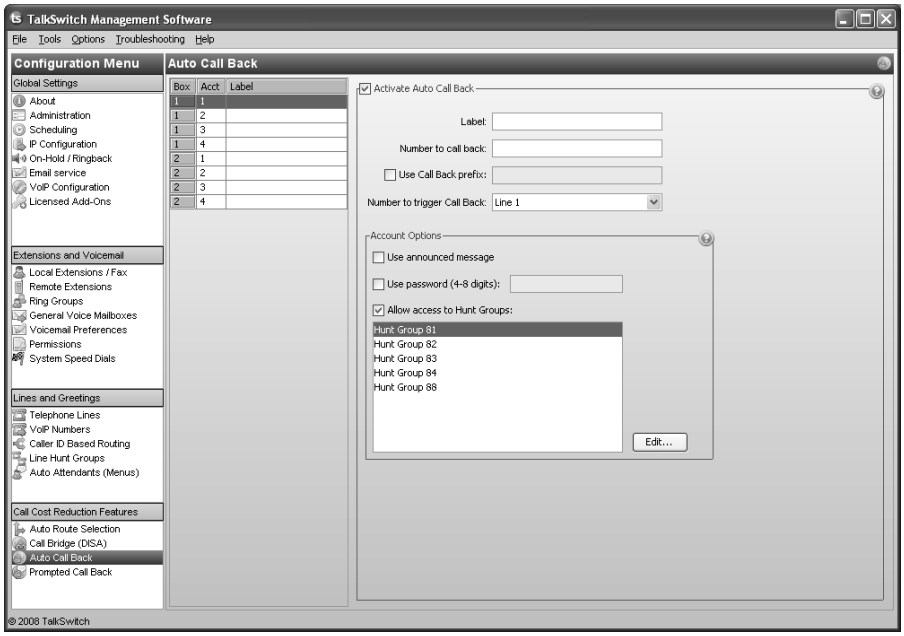
AUTO CALL BACK

Auto call back lets a travelling user gain access to the TalkSwitch system without completing the call to the telephone line or VoIP number. This minimizes long-distance and hotel telephone costs. You can set up four auto call back accounts per TalkSwitch unit.

The **Auto Call Back** page allows you to set up auto call back accounts. You can set up four auto call back accounts per TalkSwitch unit. Each auto call back account includes the name, call back number, telephone line or VoIP number that will trigger auto call back, announced message, password, dialing prefix, and hunt groups.

You must ensure the auto attendant for the telephone line or VoIP number is set to answer after four rings. In order to trigger auto call back, the user must call the telephone line or VoIP number, and then hang up after two rings, but before the auto attendant answers.

1. Select the **Auto Call Back** page.



Using auto call back

1. Call the telephone line or VoIP number, let it ring twice, and then hang up. The TalkSwitch unit will dial the call back number configured for that line.
2. Press **#** to accept the call back.
3. Enter the auto call back password + **#** to enter the system menu. From the system menu you can:
   - Dial an extension. After you complete the call, you will be disconnected.
   - Check your voicemail by dialing ** + extension. After you check your voicemail, press * to return to the system menu.
   - Call an outside number by dialing **9** (**0** in some regions), or **81**–**88** + phone number. To complete the call, press **#*** to return to the system menu. You can also press **##** to call another outside number using the same hunt group.
   - Configure the system by dialing **#**, entering the system password + **#**, and then entering the command.

When the system returns your call, it can play a default message that tells the recipient to press **#** to accept the auto call back. You can record and enable an announced message that will be played instead of the default message. For example, the announced message can tell the front desk at your hotel to transfer the call to your room. You can also disable the announced message to use the default message.

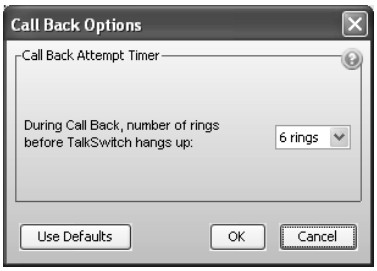
To configure auto call back, call the telephone line or VoIP number, and let the auto attendant answer. Dial **6**, and then follow the prompts. You can change the call back number, change the announced message, enable/disable the announced message, and enable/disable the auto call back account.

Activate auto call back

The *Activate Auto Call Back* area allows you to set up an auto call back account, with the name, call back number, and the telephone line or VoIP number that will trigger auto call back.

1. Select an available auto call back account.
2. Select the *Activate Auto Call Back* checkbox.
3. Enter the name of the auto call back account in the *Label* box.
4. Enter the call back number in the *Number to call back* box. This is the telephone number at the traveller's destination. Do not enter a hunt group number. Note that the travelling user can change the call back number. The new call back number will replace that entered into the **Auto Call Back** page.
5. To have the TalkSwitch unit dial a number before the call back number, select the *Use call back prefix* checkbox, and then enter the prefix. The prefix can include numbers 0–9, *****, **#**, and "," (comma for 2-second pause). For example, you can enter a calling card number, a pause, a PIN number, and another pause.
6. Select the telephone line or VoIP number in the *Number to trigger call back* list. Each auto call back account can be associated with one telephone line or VoIP number. Calling this telephone line or VoIP number and then hanging up will trigger the call back.
7. After the TalkSwitch unit dials an auto call back number or the prompted call back number, it will wait for a configurable number of rings for someone to answer. By default, it will wait six rings before hanging up. You can change the number of rings. See *Call Back* on page 190.

   a) Choose **Options > Call Back**. The **Call Back Options** window appears.



   b) Select the number of rings the TalkSwitch unit should wait before hanging up.

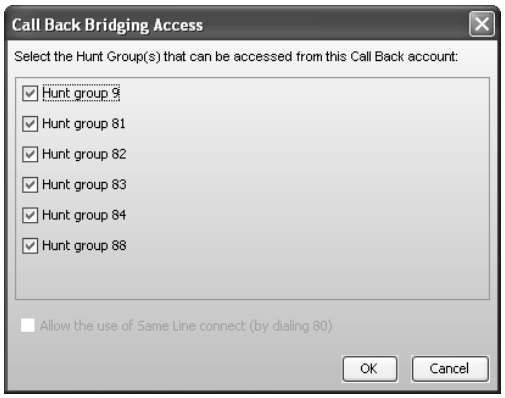
Account options

The *Account Options* area allows you to set up the announced message, password and dialing prefix.

1. To have the TalkSwitch unit play a recorded message when the call back is answered, select the *Use announced message* checkbox. This is required if a receptionist will answer the phone after the TalkSwitch unit dials the call back number. For example, the announced message can tell the hotel front desk: *"Please transfer this call to Bob in room 307"*. You can record the announced message using a local extension connected to the TalkSwitch unit, or remotely by calling a telephone line connected to the TalkSwitch unit. See *Recording an announced message* on page 170.
2. To prevent unauthorized use of the TalkSwitch system, select the *Use password* checkbox, and then enter a numeric password. The user will be required to enter this password before configuring call back, and after answering call back.

3. To select the hunt groups the user can dial, select the *Allow access to hunt groups* checkbox. This way the user can accept the call back, and then place an outbound call through the TalkSwitch system. The window enables the **Edit** button.

4. Click **Edit**. The **Call Back Bridging Access** window appears.


Call Back Bridging Access
Select the Hunt Group(s) that can be accessed from this Call Back account:
Hunt group 9
Hunt group 81
Hunt group 82
Hunt group 83
Hunt group 84
Hunt group 88
Allow the use of Same Line connect (by dialing 80)
OK
Cancel

Note: Depending on the region, callers can access either Hunt group 9 or Hunt group 0.

5. Select the hunt groups that can be used with the auto call back account.

6. If *3-Way Calling/Conference* is selected in the **Telephone Lines** page, the **Call Back Bridging Access** window enables the *Allow the use of same line connect (by dialing 80)* checkbox. Select the checkbox to enable hunt group 80.

   If the user dials hunt group 80, the TalkSwitch system will use the same telephone line to call the third party. It will have the telephone company put the user on hold. It will then dial the third party's phone number, using the same telephone line. Once the third party answers, the TalkSwitch system will connect the user and the third party.

Recording an announced message

This procedure describes how to record the announced message played by the TalkSwitch unit when the call back is answered.

1. Pick up a local extension connected to the TalkSwitch unit, or remotely call a telephone line or VoIP number connected to the TalkSwitch unit. You must record the announced message on the same unit that has the auto call back account.

2. Enter command mode by either pressing **#** on an analog extension phone, or ***55#** on a TalkSwitch IP phone (note: other brands may use ***55 Send** or ***55 Dial**).

3. Enter the system password, followed by **#**.

4. Enter the command listed below, and follow the prompts.

Auto call back account number	Record message	Erase message	Play message
1	**61 #**	**061 #**	**71 #**
2	**62 #**	**062 #**	**72 #**
3	**63 #**	**063 #**	**73 #**
4	**64 #**	**064 #**	**74 #**

5. Press ***** to exit command mode.

You can also record, enable or disable an announced message after dialing **6** at the auto attendant.

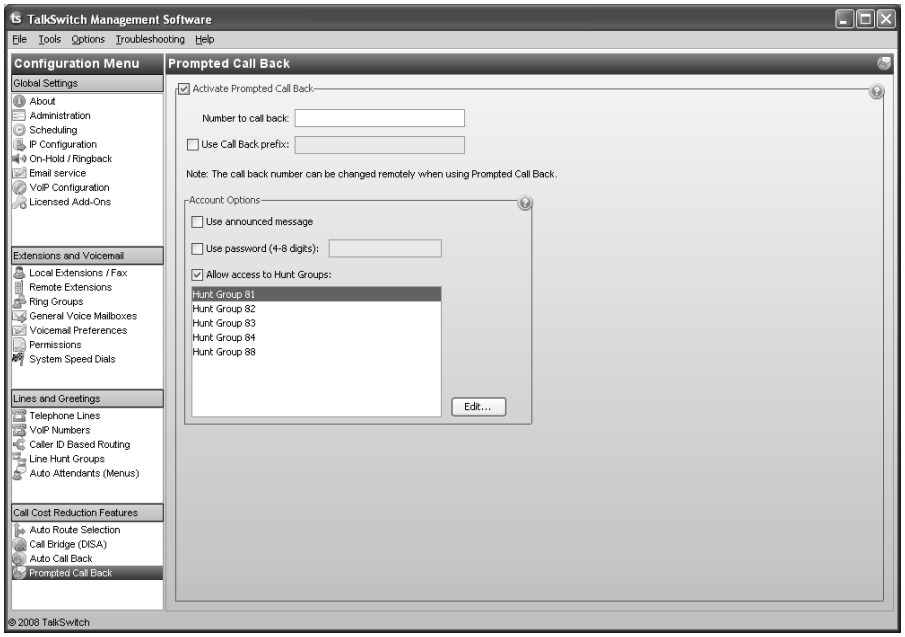
PROMPTED CALL BACK

Prompted call back lets a travelling user gain access to the TalkSwitch system. This minimizes long-distance and hotel telephone costs. You can set up one prompted call back account for the system.

The **Prompted Call Back** page allows you to set up prompted call back for the TalkSwitch system, including the name, call back number, announced message, password, dialing prefix, and hunt groups.

You must ensure an auto attendant will answer. In order to trigger prompted call back, the user must call the system, wait for the auto attendant to answer, dial **61**, and then hang up.

1. Select the **Prompted Call Back** page.



Using prompted call back

1. Call the TalkSwitch system, and let the auto attendant answer.
2. Dial **61**, and then hang up. The system will dial the call back number.
3. Press **#** to accept the call back.
4. Enter the prompted call back password + **#** to enter the system menu. From the system menu you can:
   - Dial an extension. After you complete the call, you will be disconnected.
   - Check your voicemail by dialing ** + extension. After you check your voicemail, press * to return to the system menu.
   - Call an outside number by dialing **9** (**0** in some regions), or **81**–**88** + phone number. To complete the call, press **#*** to return to the system menu. You can also press **##** to call another outside number using the same hunt group.

- Configure the system by dialing **#**, entering the system password + **#**, and then entering the command.

When the system returns your call, it can play a default message that tells the recipient to press **#** to accept the prompted call back. You can record and enable an announced message that will be played instead of the default message. For example, the announced message can tell the front desk at your hotel to transfer the call to your room. You can also disable the announced message to use the default message.

To configure prompted call back, call the TalkSwitch system, and let the auto attendant answer. Dial **6**, and then follow the prompts. You can change the call back number, change the announced message, and enable/disable the announced message.

You may incur long-distance and hotel telephone charges each time you access prompted call back from a remote location.

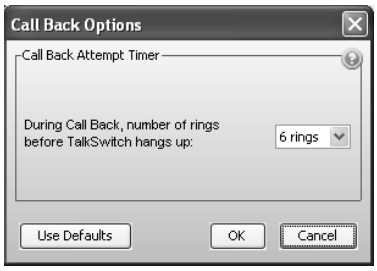
Activate prompted call back

The *Activate Prompted Call Back* area allows you to enter the name and call back number.

1. Select the *Activate Prompted Call Back* checkbox.
2. Enter the call back number in the *Number to call back* box. This is the telephone number at the traveller's destination. Do not enter a hunt group number.

   Note that the travelling user can change the call back number. The new call back number will replace that entered into the **Prompted Call Back** page.
3. To have the TalkSwitch unit dial a number before the call back number, select the *Use call back prefix* checkbox, and then enter the prefix. The prefix can include numbers **0**–**9**, *****, **#**, and "**,**" (comma for 2-second pause). For example, you can enter a calling card number, a pause, a PIN number, and another pause.
4. After the TalkSwitch unit dials an auto call back number or the prompted call back number, it will wait for a configurable number of rings for someone to answer. By default, it will wait six rings before hanging up. You can change the number of rings. See *Call Back* on page 190.

   a) Choose **Options > Call Back**. The **Call Back Options** window appears.



   b) Select the number of rings the TalkSwitch unit should wait before hanging up.

Account options

The *Account Options* area allows you to set up the announced message, password and dialing prefix.

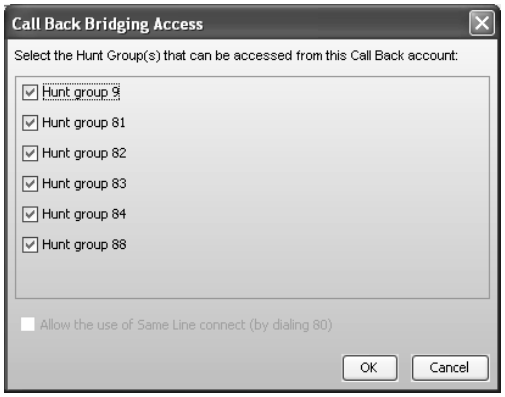
1. To have the TalkSwitch unit play a recorded message when the call back is answered, select the *Use announced message* checkbox. This is required if a receptionist will answer the phone after the TalkSwitch unit dials the call back number. For example, the announced message can tell the hotel front desk: *"Please transfer this call to Bob in room 307"*. You can

record the announced message using a local extension connected to the TalkSwitch unit, or remotely by calling a telephone line connected to the TalkSwitch unit. See *Recording an announced message* on page 173.

2. To prevent unauthorized use of the TalkSwitch system, select the *Use password* checkbox, and then enter a numeric password. The user will be required to enter this password before configuring call back, and after answering call back.

3. To select the hunt groups the user can dial, select the *Allow access to hunt groups* checkbox. This way the user can receive the call back, and then place an outbound call through the TalkSwitch system. The window enables the **Edit** button.

4. Click **Edit**. The **Call Back Bridging Access** window appears.


Call Back Bridging Access
Select the Hunt Group(s) that can be accessed from this Call Back account:
Hunt group 9
Hunt group 81
Hunt group 82
Hunt group 83
Hunt group 84
Hunt group 88
Allow the use of Same Line connect (by dialing 80)
OK
Cancel

Note: Depending on the region, callers can access either Hunt group 9 or Hunt group 0.

5. Select the hunt groups that can be used with prompted call back.

6. If *3-Way Calling/Conference* is selected in the **Telephone Lines** page, the **Call Back Bridging Access** window enables the *Allow the use of same line connect (by dialing 80)* checkbox. Select the checkbox to enable hunt group 80.

   If the user dials hunt group 80, the TalkSwitch system will use the same telephone line to call the third party. It will have the telephone company put the user on hold. It will then dial the third party's phone number, using the same telephone line. Once the third party answers, the TalkSwitch system will connect the user and the third party.

Recording an announced message

This procedure describes how to record the announced message played by the TalkSwitch unit when the call back is answered.

1. Pick up a local extension, or remotely call the TalkSwitch system.

2. Enter command mode by either pressing **#** on an analog extension phone, or ***55#** on a TalkSwitch IP phone (note: other brands may use ***55 Send** or ***55 Dial**).

3. Enter the system password, followed by **#**.

4. Enter **65#** to record the message, **065#** to erase the message, or **75#** to play the message, and then follow the prompts.

5. Press ***** to exit command mode.

You can also record, enable or disable an announced message after dialing **6** at the auto attendant.

FILE MENU

The **File** menu has commands for working with configuration files. A configuration file contains parameter settings for the TalkSwitch system and has the .tsd extension. The TalkSwitch unit uses parameter settings to control operation. The default location for storing configuration files is **C:\Program Files\TalkSwitch\TalkSwitch Configuration 6.12\Config Files**.

The commands available within the **File** menu change depending upon whether a configuration is open.

If no configuration is open, and the **Configuration Selection** page is displayed, you can:

- Create a new configuration file from the template, with default parameter settings for your TalkSwitch system. This way you can set up a configuration without a TalkSwitch unit.
- Open the configuration from a TalkSwitch unit.
- Open an existing configuration file.

If a configuration is open, you can:

- Save the configuration to a TalkSwitch unit. The TalkSwitch unit can be local or at a remote location. This is only available if you opened a configuration file.
- Save a new configuration file.
- Save a configuration to an existing configuration file, overwriting it.

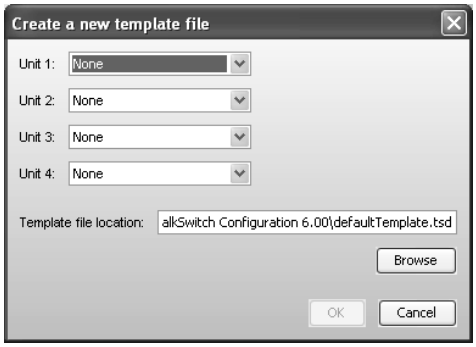
For more information about the **Configuration Selection** page, see *Starting the TalkSwitch management software* on page 22.

New Template

The **New Template** command creates a configuration file from the template, with default parameter settings for your TalkSwitch system. You can then modify the file using the TalkSwitch management software.

1. Choose **File > New Template**. The **Create a New Template File** window appears.

   Alternatively, you can click the **Open a Configuration File or Template** button in the **Configuration Selection** page, and then click the **Create a Template File** button.



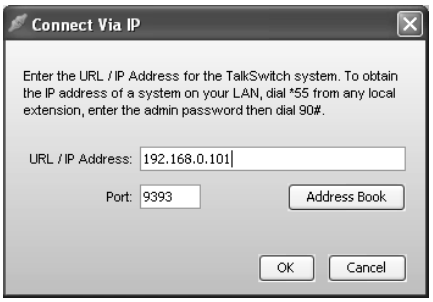
2. Select the types of TalkSwitch units in your system.
3. Click the **Browse** button to select the location and name of the configuration file.
4. Click **OK**. The system creates the file, and opens it in the TalkSwitch management software.

Open › TalkSwitch Location

The **Open > TalkSwitch Location** command connects to the selected TalkSwitch unit and retrieves the configuration. The TalkSwitch unit can be local or at a remote location. You can then modify the file using the TalkSwitch management software. The command is only available when no configuration is open.

1. Choose **File > Open > TalkSwitch Location**. The **Connect Via IP** window appears.

   Alternatively, you can also click the **Connect to a Different System** button or the **Connect to a System Via IP** button in the **Configuration Selection** page.


Connect Via IP
Enter the URL / IP Address for the TalkSwitch system. To obtain the IP address of a system on your LAN, dial *55 from any local extension, enter the admin password then dial 90#.
URL / IP Address:
192.168.0.101
Port:
9393
Address Book
OK
Cancel

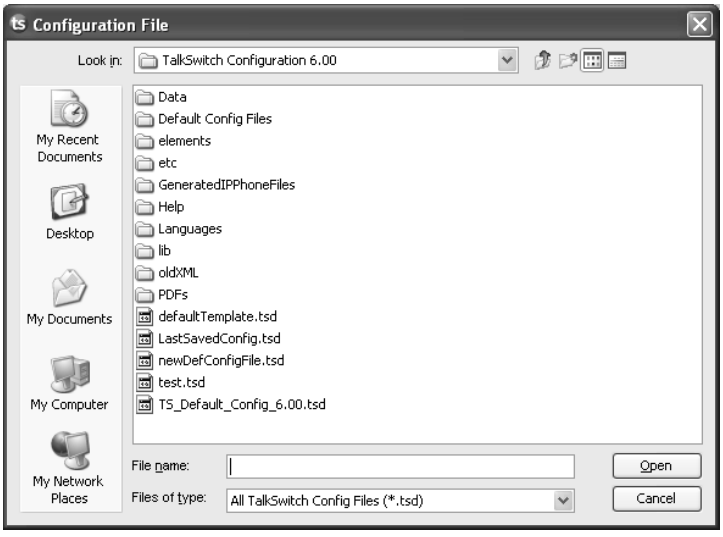
2. Enter the *URL / IP Address* of the TalkSwitch unit, and then click **OK**. You can also choose the TalkSwitch system from the address book.

Open › Configuration File

The **Open > Configuration File** command opens a saved configuration file. You can then use the TalkSwitch management software to modify the configuration. The command is only available when no configuration is open.

1. Choose **File > Open > Configuration File**. The **Configuration File** window appears.

   Alternatively, you can also click the **Open a Configuration File or Template** button in the **Configuration Selection** page, and then click the **Open a File** button.



2. Choose the configuration file, and then click **Open**.

Save

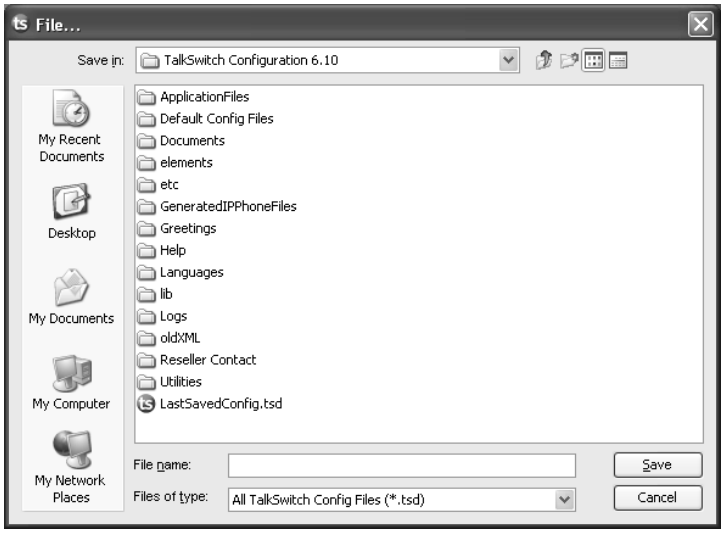
The **Save** command saves the configuration to the open configuration file or to the connected TalkSwitch unit. The TalkSwitch unit can be local or at a remote location.

NOTE: New settings must be saved to the TalkSwitch unit to take effect.

Save To › File

The **Save To > File** command saves the configuration to a new configuration file, or overwrites an existing configuration file. It does not save the parameter settings to the TalkSwitch unit.

1. Choose **File > Save To > File**. The **File** window appears.



2. Enter the name of the configuration file, and then click **Save**.

Save To › Location

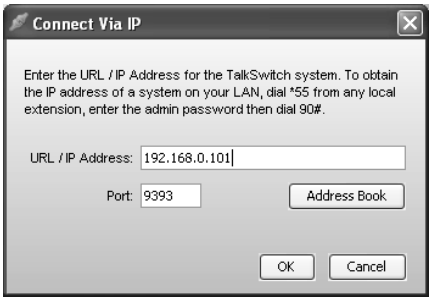
The **Save To > Location** command saves the configuration from the open configuration file to the selected TalkSwitch system. The TalkSwitch system can be local or at a remote location. The command is only available if you opened a configuration file.

1. Choose **File > Save To > Location**. The **Location** window appears. If the TalkSwitch management software auto-discovered your local TalkSwitch system, the window shows the TalkSwitch unit configured as proxy. If the TalkSwitch unit is connected by Ethernet, the window shows its IP address. If the TalkSwitch unit is connected by USB, the window shows its MAC address.

2. To save the configuration to the auto-discovered local TalkSwitch system, select the TalkSwitch unit configured as proxy, and then click **OK**.

3. To save the configuration to a remote TalkSwitch system:

   a) Click **Other TalkSwitch**. The **Connect Via IP** window appears.


Connect Via IP
Enter the URL / IP Address for the TalkSwitch system. To obtain the IP address of a system on your LAN, dial *55 from any local extension, enter the admin password then dial 90#.
URL / IP Address:
192.168.0.101
Port:
9393
Address Book
OK
Cancel

   b) Enter the *URL / IP address* of the remote TalkSwitch system, and then click **OK**. You can also choose the TalkSwitch system from the address book.

Retrieve Settings

The **Retrieve Settings** command opens the configuration from the current TalkSwitch unit. You can then modify the configuration using the TalkSwitch management software.

Alternatively, you can also click the **Configure Auto-Detected TalkSwitch System** button or the **Retry Auto Discovery** button in the **Configuration Selection** page.

Close

The **Close** command closes the configuration, and returns to the **Configuration Selection** page.

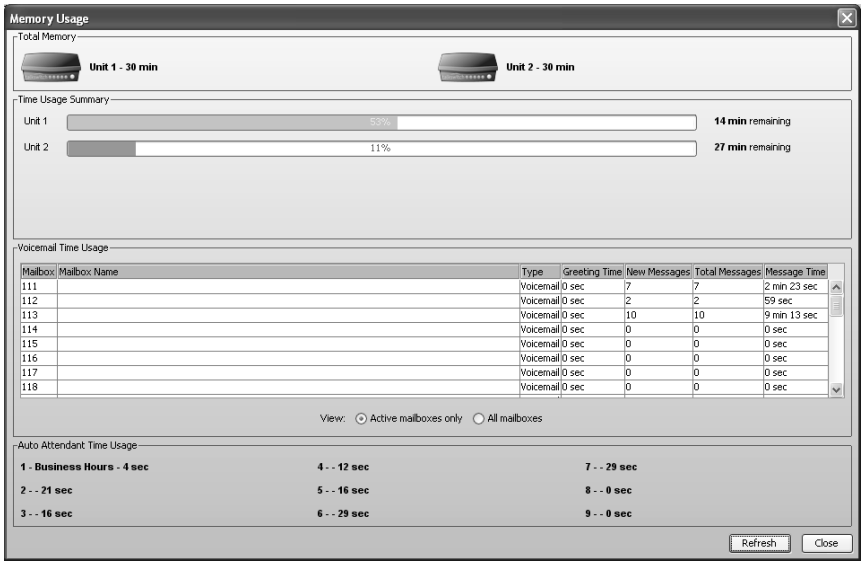
Exit

The **Exit** command closes the TalkSwitch management software.

TOOLS MENU

Memory Usage

1. Choose **Tools > Memory Usage**. The **Memory Usage** window appears. It shows the amount of time provided by the TalkSwitch units, and the amount of time used by voicemail and auto attendant messages.


Memory Usage
Total Memory
Unit 1 - 30 min
Unit 2 - 30 min
Time Usage Summary
Unit 1
53%
14 min remaining
Unit 2
11%
27 min remaining
Voicemail Time Usage
Mailbox
Mailbox Name
Type
Greeting Time
New Messages
Total Messages
Message Time
111 Voicemail 0 sec 7 7 2 min 23 sec
112 Voicemail 0 sec 2 2 59 sec
113 Voicemail 0 sec 10 10 9 min 13 sec
114 Voicemail 0 sec 0 0 0 sec
115 Voicemail 0 sec 0 0 0 sec
116 Voicemail 0 sec 0 0 0 sec
117 Voicemail 0 sec 0 0 0 sec
118 Voicemail 0 sec 0 0 0 sec
View: Active mailboxes only All mailboxes
Auto Attendant Time Usage
1 - Business Hours - 4 sec
4 - - 12 sec
7 - - 29 sec
2 - - 21 sec
5 - - 16 sec
8 - - 0 sec
3 - - 16 sec
6 - - 29 sec
9 - - 0 sec
Refresh
Close

The recording time for internal music on hold, voicemail, and the auto attendants is shared on the TalkSwitch unit. You can expand the built-in memory using memory cards.

Voicemail Manager > Mailbox Status

See *View mailbox data* on page 125.

Voicemail Manager > Delete Mailbox Password

See *Delete password* on page 126.

Voicemail Manager > Reset Mailboxes

See *Reset mailboxes* on page 125.

Firmware Auto Update

The **Firmware Auto Update** command was a legacy feature used by administrators of multiple TalkSwitch systems, such as resellers, distributors and integrators, to set up automatic updates of phone system firmware, prompts, and other essential files.

NOTE: This feature is no longer supported in Release 6.12.

Multiple TalkSwitch systems can automatically download new files simultaneously, then install the updates at a desired time. Syslog server logs confirm the success of each step of the update. Updated TalkSwitch systems can then be configured using the latest version of TalkSwitch Management Software.

With the auto update feature enabled, a TalkSwitch system will check a designated TFTP server for new update files every hour. If new files are available, the system will download them, and then report the status of the download to designated syslog and/or TFTP servers.

At a configured time the TalkSwitch system will update its firmware and reboot. The status of the update will be reported to the designated syslog and/or TFTP servers.

The steps required by the administrator to post the update files, and configure the TalkSwitch units, TFTP server, and Syslog server, are described in the sections below.

Posting firmware update files on your server

After initial set-up (see sections below), updates will occur automatically whenever the administrator posts new files on the TFTP server. To do so, follow these steps:

1. Download the latest firmware update compressed folder from the TalkSwitch website for your region. To download, select the link below according to your region. The filename indicates the firmware version number for easy reference (e.g. t611cv42_tftp.zip).

   United States and Canada

   Mexico

   South America

   Australasia

   United Kingdom and Ireland

   Mainland Europe

   Middle East and North Africa

   South Africa

   Asia-Pacific

2. Extract the files in the folder to the appropriate directory or subdirectory on your TFTP server. The contents are as follows:

   - README.TXT — Describes how to set up and use the TFTP firmware update feature, lists the contents of the compressed file and provides instructions regarding how to edit the INDEX.XML file to suit your needs.
   - INDEX.XML — Lists information about the files in a format used by TalkSwitch systems to trigger and manage updates.
   - BBLOCK — The boot block is essential for proper startup of a TalkSwitch system.
   - FIRMWARE — The program that manages all TalkSwitch phone system features.
   - ADSI — Updates ADSI information on the TalkSwitch unit (this file rarely changes).
   - TALKSET — Contains firmware for the TalkSwitch TS-200 phone (this file rarely changes).
   - ENGLISH.TLP — American English prompt file.
   - BRITISH.TLP — International English prompt file.
   - FRENCH.TLP — French prompt file.
   - SPANISH.TLP — Spanish prompt file.

   Note: TalkSwitch units will only download prompt (.TLP) files for languages previously installed on the phone system.

3. If you need to suppress distribution of certain files, and/or alter which standard reboot option the TalkSwitch unit will use after installing a firmware update, you will need to create a new version of the INDEX.XML file. To do this:

   a) Make a backup copy of INDEX.XML and rename it INDEX.BAK.

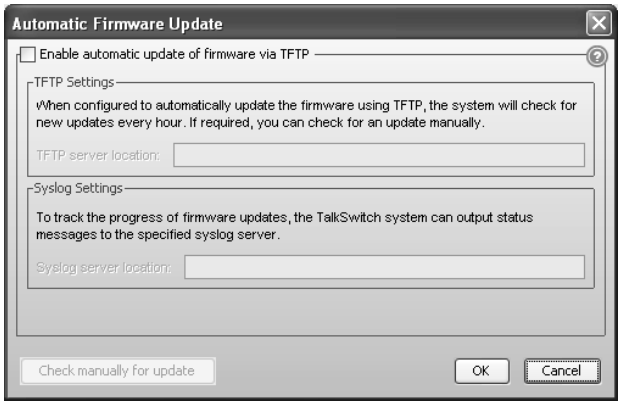
   b) Edit INDEX.XML according to the instructions in the README.TXT file. When the edits are complete SAVE them.

   By default, TalkSwitch systems configured with the firmware auto update feature will automatically download new files within an hour. Updates to language prompts take effect immediately. If changes have been made to essential firmware, the unit(s) will reboot according to the settings in the INDEX.XML file (by default at 3:00 a.m. local time).

Configuring TalkSwitch systems

To have TalkSwitch system(s) automatically download new update files, perform the following steps on each system:

1. Connect to the TalkSwitch system that you wish to automatically update using the TalkSwitch Management Software. Choose **Tools > Firmware Auto Update**. The **Automatic Firmware Update** window appears.


Automatic Firmware Update
Enable automatic update of firmware via TFTP
TFTP Settings
When configured to automatically update the firmware using TFTP, the system will check for new updates every hour. If required, you can check for an update manually.
TFTP server location:
Syslog Settings
To track the progress of firmware updates, the TalkSwitch system can output status messages to the specified syslog server.
Syslog server location:
Check manually for update
OK
Cancel

2. Select *Enable automatic update of firmware via TFTP*.

3. In the *TFTP Settings* > *TFTP server location* field, enter the IP address or FQDN of the TFTP server that has the firmware files. You can also specify a subdirectory after the address (e.g. 123.45.67.89/subdirectory1). This way you can set up separate subdirectories to provide different firmware files to different clients.

   Instructions for setting up a TFTP server are listed below.

4. In the *Syslog Settings* > *Syslog server location* field, enter the IP address or FQDN of the syslog server that will log the status of TalkSwitch firmware updates. By default TalkSwitch uses port 514 to transmit Syslog events. If you wish to use an alternate port, you can specify it by appending a colon followed by the desired port number (e.g. 123.45.67.89:555).

   Instructions for setting up a Syslog server are listed below.

5. By default the TalkSwitch system will check for updates every hour. To have the TalkSwitch system check for new updates immediately, click the **Check manually for update** button. Though no progress indication is provided by the TalkSwitch management software, update events will be reported on the syslog server.

Setting up a TFTP server

1. Download and install TFTP software to a server accessible by all TalkSwitch units that you will manage. Note the IP address or FQDN of the server: it will be needed to configure the TalkSwitch units you administer.

   One TFTP application (free) that can support simultaneous (multi-thread) downloads can be found at Solarwinds **www.solarwinds.com**.

2. Configure the TFTP settings according to the documentation provided by the supplier of your TFTP software.

3. If needed, set up subdirectories for each set of clients who require different versions of firmware.

4. If your TFTP server uses the Windows Vista operating system, you must configure the firewall by enabling port 69 as an exception. To do this follow these steps:

   a) Right-click the **Network** icon on the desktop, and then select **Properties**.

   b) Select **Windows Firewall**.

   c) Select **Change Settings**.

   d) Select the *Exceptions* tab.

   e) If the TFTP server application has already added an exception to the list, select the exception.

   f) If the TFTP server application has not added an exception to the list, then add the TFTP port. Use **TFTP** as the name, **69** as the port and **UDP** as the protocol.

   g) Save the changes.

5. Download new TalkSwitch update files and extract them to the appropriate directory on the TFTP server (typically the root directory). Edit the INDEX.XML file to suit your needs. See the section above for detailed instructions. By default, all TalkSwitch systems configured with the firmware auto update feature will automatically download the new firmware files within an hour.

Setting up a syslog server

1. Download and install Syslog software to a server accessible by all TalkSwitch units that you will manage, preferably the same device hosting your TFTP server. Note the IP address or FQDN of the server: it will be needed to configure the TalkSwitch units you administer.

   One Syslog application (free) can be found at Kiwi Enterprises **www.kiwisyslog.com**.

2. Configure the Syslog daemon settings according to the documentation provided by the supplier of your Syslog software. By default, TalkSwitch uses port 514 to transmit Syslog events.

Terminal Window (CLI)

The **Terminal Window (CLI)** command displays the **TalkSwitch Command Console** window. Use this as directed by TalkSwitch customer support.

Syslog Output

The **Syslog Output** command allows you to set up logging of system status information. Use this as directed by VoIP service providers.

Call Logging Output (CDR)

The **Call Logging Output (CDR)** command allows you to set up logging of call detail records. For information about call logging, see *Enabling call detail record logging* on page 226.

Defaults › Entire Configuration

The **Defaults > Entire Configuration** command resets the entire configuration to defaultvalues.

Defaults › Current Page

The **Defaults > Current Page** command resets the current page of the configuration to default values.

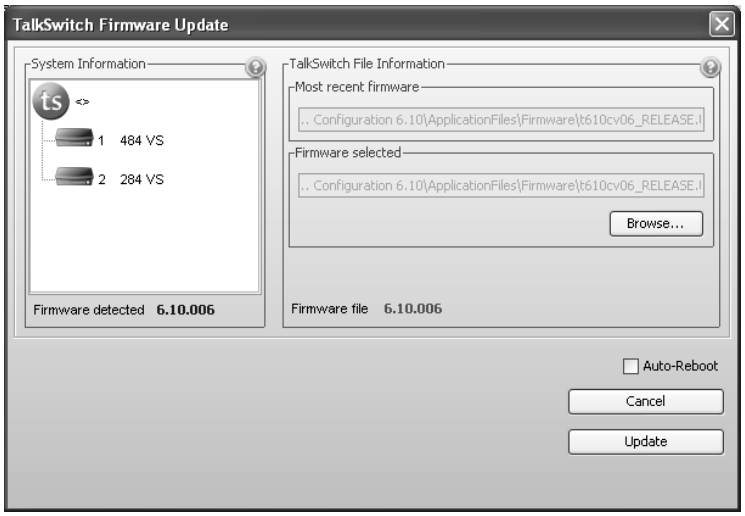
Check for Updates

The **Check for Updates** command looks for updates to the firmware or TalkSwitch management software on the TalkSwitch website.

Update Firmware

The **Update Firmware** command shows the firmware version of the TalkSwitch units and the latest firmware version available with the TalkSwitch management software. It allows you to update the TalkSwitch units with a newer firmware version.

1. Choose **Tools > Update Firmware**. The **TalkSwitch Firmware Update** window appears.



2. If the *Firmware detected* is lower than the *Firmware file*:

   a) Set the *Auto-Reboot* checkbox.

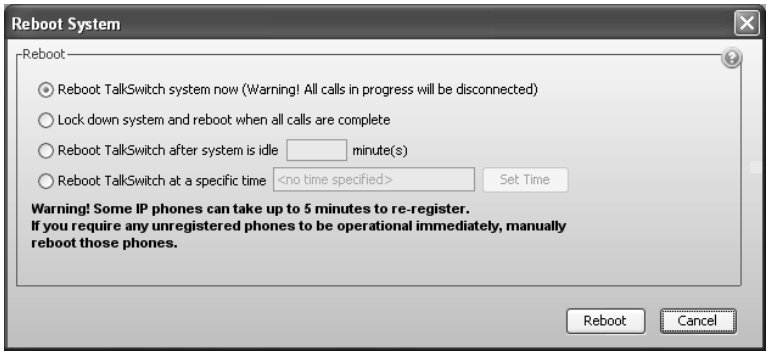
   - Select *Auto-Reboot* to automatically reboot the TalkSwitch units once the firmware has been updated. This will cancel any calls in progress.
   - Clear *Auto-Reboot* to prevent the TalkSwitch units from rebooting automatically. You will have to manually reboot the TalkSwitch units later, when there are no calls in progress. See *Reboot TalkSwitch* on page 183.

   b) Click the **Update** button.

Reboot TalkSwitch

The **Reboot TalkSwitch** command reboots the TalkSwitch units at the specified time. This is required after updating the firmware, unless you selected the *Auto-Reboot* checkbox, and the TalkSwitch units rebooted automatically.

1. Choose **Tools > Reboot TalkSwitch**. The **Reboot System** window appears.


Reboot System
Reboot
Reboot TalkSwitch system now (Warning! All calls in progress will be disconnected)
Lock down system and reboot when all calls are complete
Reboot TalkSwitch after system is idle
minute(s)
Reboot TalkSwitch at a specific time
<no time specified>
Set Time
Warning! Some IP phones can take up to 5 minutes to re-register.
If you require any unregistered phones to be operational immediately, manually reboot those phones.
Reboot
Cancel

2. Select when the TalkSwitch units should reboot. They can reboot immediately, when all calls are complete, once the system has been unused for a period of time, or once a certain time arrives.

3. Click **Reboot**.

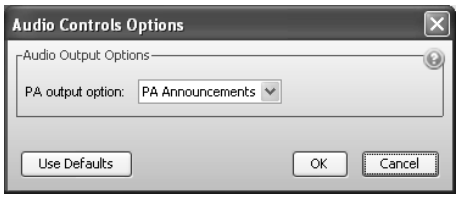
OPTIONS MENU

The **Options** menu has commands for configuring the system.

PA Output

The **PA Output** command allows you to set the audio from the PA jack.

1. Choose **Options > PA Output**. The **Audio Control Options** window appears.



2. Select the audio that is output from the PA jack. Choices are:

   - *PA announcements* — Routes a PA announcement to the PA jack when a user dials ***0**.

     If a user has dialed ***80** to route music on hold through the PA jack, and a user dials ***0** to make a PA announcement, the PA announcement will interrupt the music on hold.

   - *Voicemail screening* — Routes a call to the PA jack if a caller reaches a voice mailbox, or if a user accesses a voice mailbox. In addition, the system will route a PA announcement to the PA jack if a user dials ***0**.

     NOTE: Unlike a PA announcement, voicemail screening will not interrupt music on hold.

   - *Demo with router* — Intended for TalkSwitch demonstration purposes only.

   - *Static VoIP demo* — Intended for TalkSwitch demonstration purposes only.

Transfer Preferences

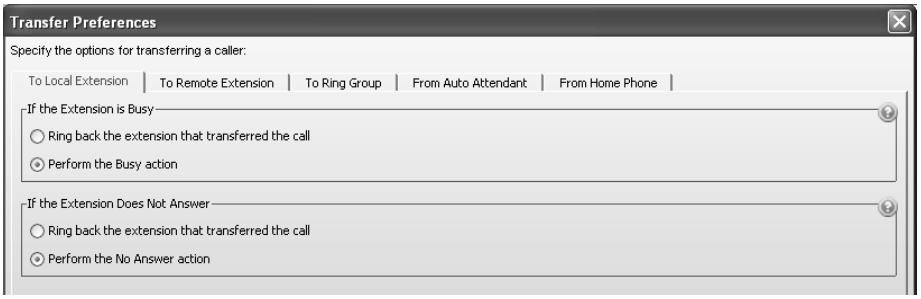
The **Transfer Preferences** command allows you to set options for transferring callers. You can set options for calls transferred to local extensions, remote extensions and ring groups, and from auto attendants and home phones.

1. Choose **Options > Transfer Preferences**. The **Transfer Preferences** window appears.

To local extension tab

The *To Local Extension* tab allows you to set call handling for calls manually transferred to a local extension, if the extension is busy or unanswered.

1. Select the *To Local Extension* tab.


Transfer Preferences
Specify the options for transferring a caller:
To Local Extension
To Remote Extension
To Ring Group
From Auto Attendant
From Home Phone
If the Extension is Busy
Ring back the extension that transferred the call
Perform the Busy action
If the Extension Does Not Answer
Ring back the extension that transferred the call
Perform the No Answer action

2. Select the option if the extension is busy:

   - *Ring back extension that transferred the call* — If the extension is busy, transfer the call back to the source extension.
   - *Perform Busy at extension action* — If the extension is busy, perform the target extension's Busy call cascade.

3. Select the option if the extension is unanswered:

   - *Ring back extension that transferred the call* — If the extension is unanswered, transfer the call back to the source extension.
   - *Perform No Answer at extension action* — If the extension is unanswered, perform the target extension's No Answer call cascade.

To remote extension tab

The *To Remote Extension* tab allows you to set call handling for calls manually transferred to a remote extension, if the extension is busy or unanswered. It also sets transfer and clear, and screened or blind transfer.

1. Select the *To Remote Extension* tab.

Transfer Preferences
Specify the options for transferring a caller:
To Local Extension
To Remote Extension
To Ring Group
From Auto Attendant
From Home Phone
If the extension is busy
Ring back the extension that transferred the call
Perform the Busy action
If the extension does not answer
Ring back the extension that transferred the call
Perform the No Answer action
Transfer and Clear
Clear telephone line after call transferred
Remote Transfer
When transferring calls from a remote extension:
Allow screening of calls
Perform a blind transfer

2. Select the option if the remote extension is busy:

   - *Ring back extension that transferred the call* — If the remote extension is busy, transfer the call back to the source extension.

   - *Perform Busy at extension action* — If the remote extension is busy, perform the remote extension's Busy call cascade.

3. Select the option if the remote extension is unanswered:

   - *Ring back extension that transferred the call* — If the remote extension is unanswered, transfer the call back to the source extension.

   - *Perform No Answer at extension action* — If the remote extension is unanswered, perform the remote extension's No Answer call cascade.

4. Select the *Clear telephone line after call transferred* checkbox to clear the telephone line after the call is transferred to a remote extension. The telephone line must have the Transfer and Clear service available and selected on the **Telephone Lines** page. See *Phone line services* on page 134.

5. Select how the TalkSwitch system transfers calls from a remote extension to a local extension, ring group or VoIP number within a TalkSwitch VoIP network. Choices are:

   - *Allow screening of calls* — When the user at a remote extension transfers a call, allows them to determine whether the other user wants it.

   - *Perform a blind transfer* — When the user at a remote extension transfers a call, the system immediately connects the caller to the other user and releases the telephone line connected to the remote extension.

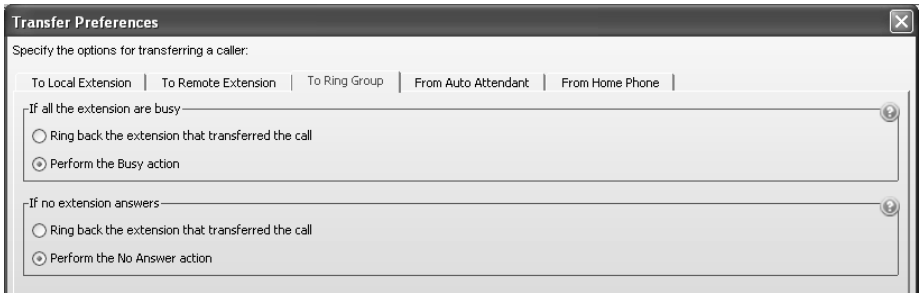
Transferring a call from a remote extension requires telephone lines without the 3-Way Calling/Conference service or the Transfer and Clear service.

When a call is transferred from one remote extension to another remote extension, it is always a blind transfer to avoid tying up a third line.

To ring group tab

The *To Ring Group* tab allows you to set call handling for calls manually transferred to a ring group, if all the extensions are busy or unanswered.

1. Select the *To Ring Group* tab.


Transfer Preferences
Specify the options for transferring a caller:
To Local Extension
To Remote Extension
To Ring Group
From Auto Attendant
From Home Phone
If all the extension are busy
Ring back the extension that transferred the call
Perform the Busy action
If no extension answers
Ring back the extension that transferred the call
Perform the No Answer action

2. Select the option if all the extensions are busy:

   - *Ring back extension that transferred the call* — If all the extensions are busy, transfer the call back to the source extension.
   - *Perform Busy at extension action* — If all the extensions are busy, perform the ring group's Busy call cascade.

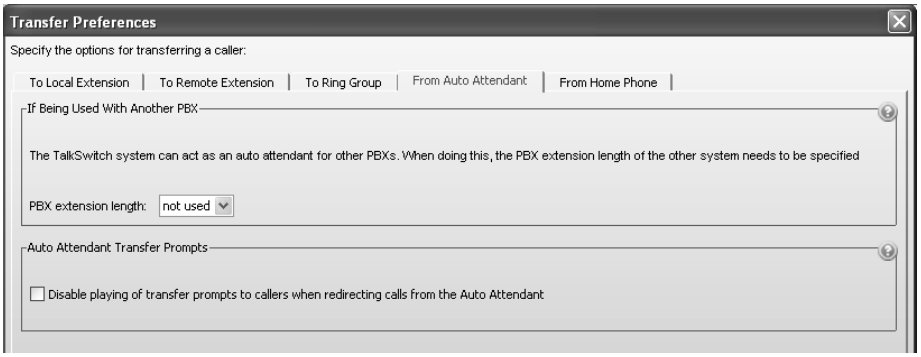
3. Select the option if all the extensions are unanswered:

   - *Ring back extension that transferred the call* — If all the extensions are unanswered, transfer the call back to the source extension.
   - *Perform No Answer at extension action* — If all the extensions are unanswered, perform the ring group's No Answer call cascade.

From auto attendant tab

The *From Auto Attendant* tab allows you to set the PBX extension length and disable the transfer prompts.

1. Select the *From Auto Attendant* tab.


Transfer Preferences
Specify the options for transferring a caller:
To Local Extension
To Remote Extension
To Ring Group
From Auto Attendant
From Home Phone
If Being Used With Another PBX
The TalkSwitch system can act as an auto attendant for other PBXs. When doing this, the PBX extension length of the other system needs to be specified
PBX extension length:
not used
Auto Attendant Transfer Prompts
Disable playing of transfer prompts to callers when redirecting calls from the Auto Attendant

2. If the TalkSwitch system is used as an auto attendant for another PBX, set the *PBX extension length* to the number of digits the PBX uses for its extensions. When the caller dials **7** followed by an extension number, the TalkSwitch unit will capture the extension number, flash the line and then dial the extension number to complete the transfer. Choices range from *Not used* to *7 digits*.

See *Connecting a TalkSwitch unit to a PBX* on page 21.

This feature only works with PBXs that use hook flash as a transfer method. Please contact your PBX manufacturer for transfer details.

3. Select the *Disable playing of transfer prompts* checkbox to disable the *"one moment please"* prompt when transferring a call.

From home phone tab

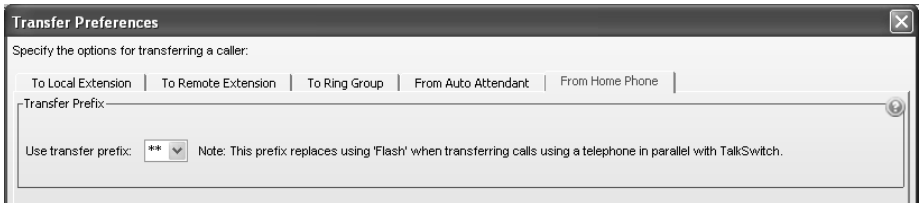
The *From Home Phone* tab allows you to set the transfer prefix. Dialing the transfer prefix indicates you will manually transfer a call from a phone connected in parallel to the TalkSwitch unit, to an extension or voice mailbox within the TalkSwitch unit. It is equivalent to pressing **Flash** or **Recall** or **Link** on a local extension.

This is required if you have a phone connected to the same telephone line as the TalkSwitch unit, but the phone is not connected to the TalkSwitch unit itself (i.e. the phone and the TalkSwitch unit are connected in parallel to the telephone line).

For example, a home-based business has one telephone line. A home phone is connected to the line through a wall jack in the kitchen. The TalkSwitch unit is connected to the same line through a wall jack in the office. Therefore the home phone is connected in parallel to the TalkSwitch unit.

Remote extension 210 is set up to route calls to the owner's cell phone. The owner lets the TalkSwitch unit answer the phone during the day, but answers the home phone at night and on weekends. If a family member answers the home phone at night while the owner is away, they can manually transfer a business call to the owner's cell phone. To do this, they dial the transfer prefix (****** or ***#**), wait for the double beep, dial the remote extension number (**210**), and then hang up. The TalkSwitch system then routes the call to the owner's cell phone.

1. Select the *From Home Phone* tab.


Transfer Preferences
Specify the options for transferring a caller:
To Local Extension
To Remote Extension
To Ring Group
From Auto Attendant
From Home Phone
Transfer Prefix
Use transfer prefix:
**
Note: This prefix replaces using 'Flash' when transferring calls using a telephone in parallel with TalkSwitch.

2. Select the transfer prefix. Choices are:

- ******
- ***#**

Call Reminders

The **Call Reminders** command allows you to set reminder timers. When a reminder timer expires, the phone rings indicating a caller is on hold or parked. In addition, a queued caller hears a prompt when the *Queue options reminder timer* expires.

1. Choose **Options > Call Reminders**. The **Call Reminder Timers** window appears.

2. Set the *Hold ringback reminder timer* to how often the phone should ring when a caller is on hold.

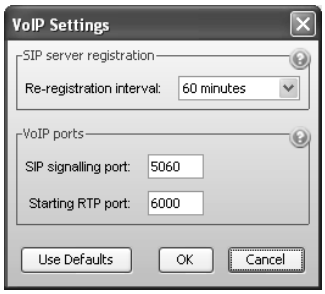
3. Set the *Park ringback reminder timer* to how often the phone should ring when a caller is parked.

4. Set the *Queue options reminder timer* to how often a queued caller should hear a prompt.

VoIP Trunking

The **VoIP Trunking** command allows you to set the SIP server registration message interval, the SIP server port, and the starting RTP port. These settings are used if this location is the SIP server for a TalkSwitch VoIP network.

1. Choose **Options > VoIP Trunking**. The **VoIP Settings** window appears.



SIP server registration

The *SIP Server Registration* area allows you to set the SIP server registration interval. This setting is used if this location is the SIP server for a TalkSwitch VoIP network.

1. Select the amount of time between registration messages from the SIP server. Use a higher number if you have a large TalkSwitch VoIP network. Choices range from *5 minutes* to *60 minutes*. The default value is *60 minutes*.

VoIP ports

The *VoIP Ports* area allows you to set the SIP server port, and the starting RTP port. These settings are used if this location is the SIP server for a TalkSwitch VoIP network.

1. The *SIP signalling port* box allows you to change the SIP server port. The default SIP server port is 5060. If required, enter a different port number ranging from 1024 to 65535.

   If your SIP server is at another location, and the SIP server uses a SIP server port other than 5060 (e.g. 5061), do not set *SIP signalling port* to that number. Instead, include the port number in the *Proxy server* box of the **System Information > VoIP Configuration** window (e.g. 222.234.432.234:5061).

2. The default *Starting RTP port* is 6000. If required, enter a different port number ranging from 1024 to 65535.

   Each TalkSwitch unit with four VoIP lines requires four RTP ports. The RTP ports are evenly numbered from the starting port as follows:

   - TalkSwitch unit 1 — 6000 to 6006
   - TalkSwitch unit 2 — 6010 to 6016
   - TalkSwitch unit 3 — 6020 to 6026
   - TalkSwitch unit 4 — 6030 to 6036

   Each TalkSwitch unit with eight VoIP lines requires eight RTP ports. The RTP ports are evenly numbered from the starting port as follows:

   - TalkSwitch unit 1 — 6100 to 6114
   - TalkSwitch unit 2 — 6200 to 6214
   - TalkSwitch unit 3 — 6300 to 6314
   - TalkSwitch unit 4 — 6400 to 6414

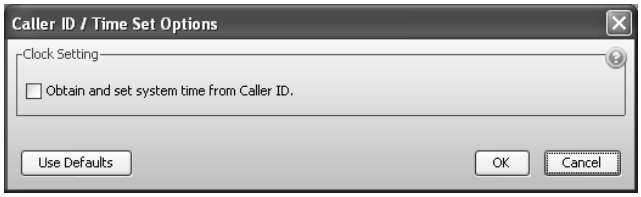
   These are the RTP ports the TalkSwitch system listens to for audio. If these are not opened correctly, users will not hear their callers.

3. Ensure your router is set up to perform port forwarding for the SIP signalling and RTP ports. See *Router port forwarding* on page 36.

Clock Preferences

The **Clock Preferences** command allows you to have the TalkSwitch system get the time and date (but not the year or time zone) from caller ID. This allows the system to refresh the clock automatically. It is beneficial if your area has frequent power outages that cause the system time to become inaccurate. However it is not recommended if you have more than one telephone service (e.g. regular telephone and VoIP) because the telephone services could have conflicting times.

1. Choose **Options > Clock Preferences**. The **Caller ID / Time Set Options** window appears.

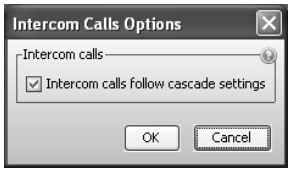


2. Select the *Obtain system time from caller ID* checkbox to have the TalkSwitch unit update its internal clock using the timestamp within caller ID. If it is not selected, set the time using the **TalkSwitch Date and Time Properties** window. See *System time* on page 31.

Internal Calls

The **Internal Calls** command allows you to set whether intercom calls use the call cascade of the dialed extension if the extension is busy, unanswered or in do not disturb mode. An intercom call is a call from a local extension to another local extension, remote extension or ring group.

1. Choose **Options > Internal Calls**. The **Intercom Calls Options** window appears.

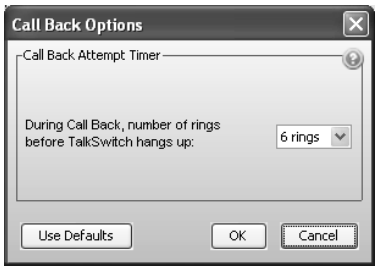


2. Select the *Intercom calls follow cascade settings* checkbox to have an intercom call use the call cascade of the dialed extension. Otherwise the intercom call will ring continuously.

Call Back

The **Call Back** command allows you to set the number of rings the TalkSwitch unit will allow before hanging up, during an auto call back or prompted call back.

1. Choose **Options > Call Back**. The **Call Back Options** window appears.



2. Select the number of rings the TalkSwitch unit will allow during a call back before hanging up. Choices range from *1 ring* to *9 rings*.

Set Date & Time

The **Set Date & Time** command allows you to set the time, date, time zone, and NTP server. See *System time* on page 31.

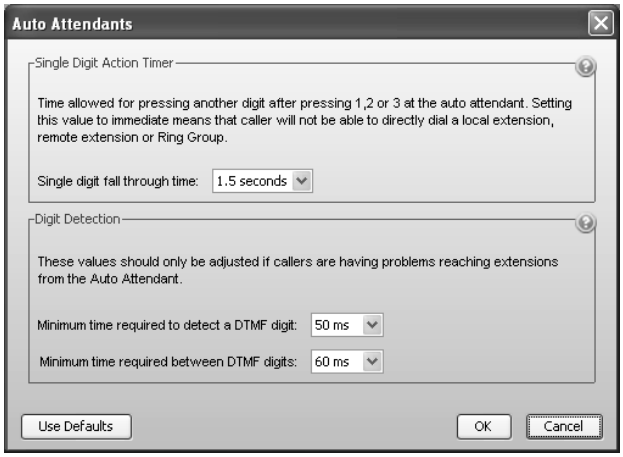
TROUBLESHOOTING MENU

The **Troubleshooting** menu allows access to diagnostics and advanced settings. Do not adjust these settings unless advised to do so by your local dealer or technical support centre. The default settings provide correct operation with telephone lines and analogue extensions.

Auto Attendants

The **Auto Attendants** command allows you to set auto attendant timers.

1. Choose **Troubleshooting > Auto Attendants**. The **Auto Attendants** window appears.


Auto Attendants
Single Digit Action Timer
Time allowed for pressing another digit after pressing 1,2 or 3 at the auto attendant. Setting this value to immediate means that caller will not be able to directly dial a local extension, remote extension or Ring Group.
Single digit fall through time:
1.5 seconds
Digit Detection
These values should only be adjusted if callers are having problems reaching extensions from the Auto Attendant.
Minimum time required to detect a DTMF digit:
50 ms
Minimum time required between DTMF digits:
60 ms
Use Defaults
OK
Cancel

2. Set the *Single digit fall through time* to how long the auto attendant will wait after the caller dials a single digit. Once the wait time elapses, the auto attendant processes the entry. Choices range from *immediately* to *4 seconds*.

   Selecting *immediately* limits each entry to a single number. The other selections allow callers to enter extension and ring group numbers.

   If the caller dials three digits, the auto attendant immediately processes the entry.

3. Select the *Minimum required time to detect a DTMF digit*. This is how long the caller must press the number key on their phone for the TalkSwitch unit to detect a valid digit. Choices range from *40 ms.* to *100 ms.*

   Adjust this if callers are misdirected after selecting options using the auto attendant. Use a shorter time if the TalkSwitch unit appears to have trouble hearing the digits. Use a longer time if the auto attendant is misdirecting callers.

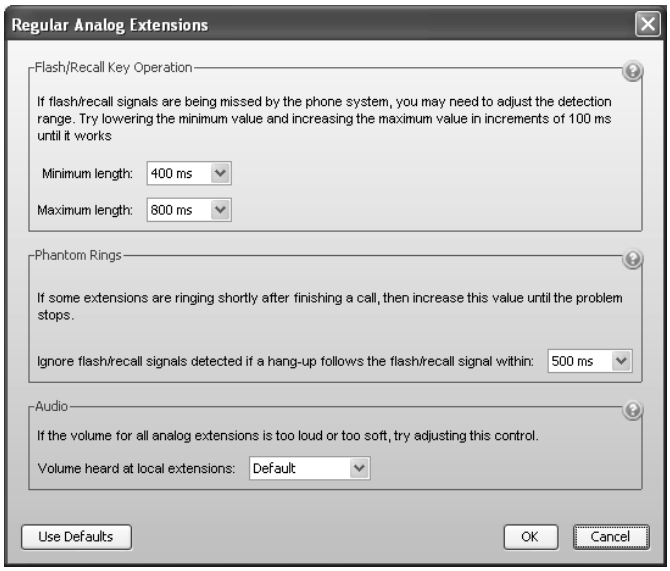
4. Select the *Minimum required time between DTMF digits.* This is how long the caller must wait after releasing the number key on their phone for the TalkSwitch unit to detect the next digit. Choices range from *30 ms.* to *80 ms.*

   Use a longer time if the auto attendant is misdirecting callers due to echoed digits from cell phones. For example, calls are misdirected to extension 111. This can happen if the caller dials a different extension number (e.g. 113), but the telephone network echoes the 1, so the TalkSwitch unit hears 1113. It captures the first three digits, ignores the 3, and misdirects the call to extension 111.

Extensions > Regular Analog Extensions

The **Extensions > Regular Analog Extensions** command allows you to set analog extension timers and the volume level.

1. Choose **Troubleshooting > Extensions > Regular Analog Extensions**. The **Regular Analog Extensions** window appears.


Regular Analog Extensions
Flash/Recall Key Operation
If flash/recall signals are being missed by the phone system, you may need to adjust the detection range. Try lowering the minimum value and increasing the maximum value in increments of 100 ms until it works
Minimum length: 400 ms
Maximum length: 800 ms
Phantom Rings
If some extensions are ringing shortly after finishing a call, then increase this value until the problem stops.
Ignore flash/recall signals detected if a hang-up follows the flash/recall signal within: 500 ms
Audio
If the volume for all analog extensions is too loud or too soft, try adjusting this control.
Volume heard at local extensions: Default
Use Defaults
OK
Cancel

2. Set the *Minimum length* of the Flash to the shortest duration considered to be an intentional Flash from a local extension. Choices range from *300 ms* to *650 ms*.

3. Set the *Maximum length* of the Flash to the longest duration considered to be an intentional Flash from a local extension. Choices range from *550 ms* to *1000 ms*.

   Most phones emit a 600 ms Flash signal when the Flash or Link button is pressed.

4. When the user hangs up, placing the phone on hook might generate false Flash signals. Set the *Ignore flash signals detected* list so the TalkSwitch system can ignore false Flash signals during a hang up. This prevents false holds and ring backs. Choices range from *400 ms* to *1 second*.

5. Set the inbound volume level for the local analog extensions. This affects the volume of calls and system prompts for internal users, not remote callers. Choices are:

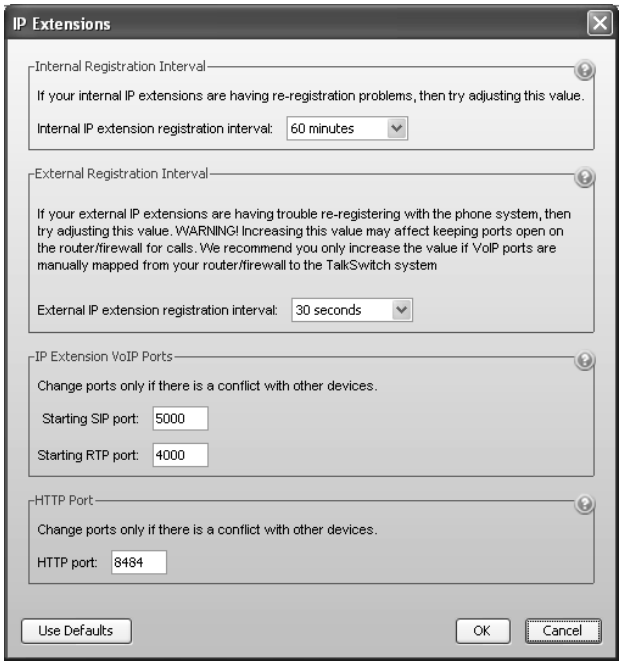
   - *Loudest*
   - *Default*
   - *Quieter*
   - *Quietest*

Extensions > IP Extensions

The **Extensions > IP Extensions** command allows you to set the amount of time between registration messages from IP extensions. You can also change the starting SIP, starting RTP and HTTP ports.

Do not adjust these settings unless advised to do so by your local dealer or technical support centre. If your system does not appear to be functioning correctly, please contact your local dealer or technical support centre for assistance.

1. Choose **Troubleshooting > Extensions > IP Extensions**. The **IP Extensions** window appears.


IP Extensions
Internal Registration Interval
If your internal IP extensions are having re-registration problems, then try adjusting this value.
Internal IP extension registration interval:
60 minutes
External Registration Interval
If your external IP extensions are having trouble re-registering with the phone system, then try adjusting this value. WARNING! Increasing this value may affect keeping ports open on the router/firewall for calls. We recommend you only increase the value if VoIP ports are manually mapped from your router/firewall to the TalkSwitch system
External IP extension registration interval:
30 seconds
IP Extension VoIP Ports
Change ports only if there is a conflict with other devices.
Starting SIP port:
5000
Starting RTP port:
4000
HTTP Port
Change ports only if there is a conflict with other devices.
HTTP port:
8484
Use Defaults
OK
Cancel

2. Select the *Internal IP extension registration interval*, which is the interval for registration messages from IP extensions within the office. Choices range from *2 minutes* to *60 minutes*.

3. Select the *External IP extension registration interval*, which is the interval for registration messages from IP extensions that are remotely located outside the office. Choices range from *15 seconds* to *60 minutes*.

4. The default *Starting SIP port* is 5000. If required, enter a different port ranging from 1024 to 65535.

5. The default *Starting RTP port* is 4000. If required, enter a different port number ranging from 1024 to 65535.

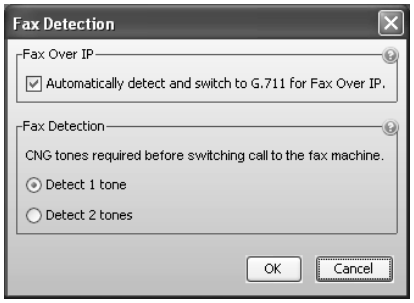
6. The default *HTTP port* is 8484. If required, enter a different port number ranging from 1024 to 65535. If your system will use a TalkSwitch IP phone as an external IP extension, ensure the router maps this port to the TalkSwitch unit acting as local proxy.

Fax Detection

The **Fax Detection** command allows you to set up detection of Fax Over IP and CNG tones.

Do not adjust these settings unless advised to do so by your local dealer or technical support centre. If your system does not appear to be functioning correctly, please contact your local dealer or technical support centre for assistance.

1. Choose **Troubleshooting > Fax Detection**. The **Fax Detection** window appears.



2. Select the *Automatically detect and switch to G.711 for fax over IP* checkbox to have the TalkSwitch unit detect fax over IP and then switch to G.711.

3. Set the number of CNG tones the TalkSwitch unit must detect before the auto attendant routes the call to the fax machine. Choices are:

   - *Detect 1 tone* — The auto attendant will route the call to the fax machine if the TalkSwitch unit detects one CNG tone. Select this if fax calls are not being routed to the fax machine.

   - *Detect 2 tones* — The auto attendant will route the call to the fax machine if the TalkSwitch unit detects two CNG tones. Select this if regular calls are being routed to the fax machine.

Telephone Lines › Audio

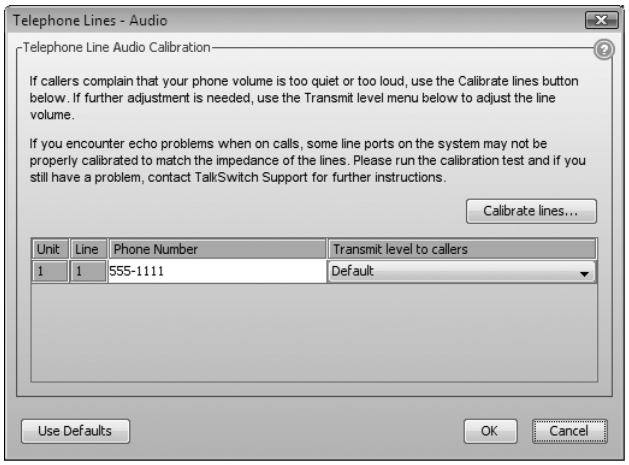
For the best call audio quality and volume levels, your TalkSwitch system parameters must be matched to your telephone lines. The *Line Optimization* area offers two calibration methods to match your telephone lines.

The **Telephone Lines > Audio** command allows you to automatically calibrate the TalkSwitch unit to match the far end impedance of the telephone lines; to select the matching near end impedance of the telephone lines; and to select the transmit level to callers.

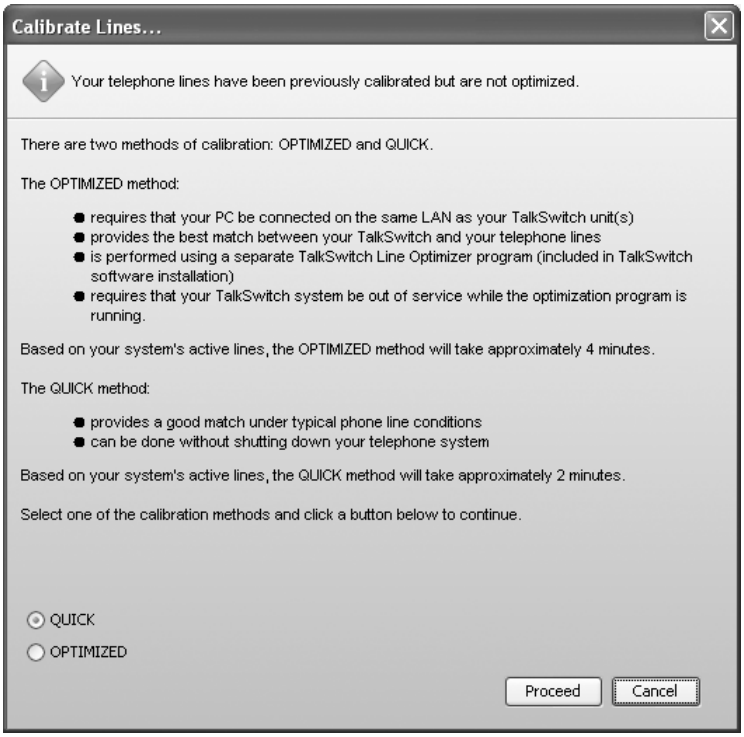
All controls on this page update the TalkSwitch unit in real time. This allows you to adjust levels without saving the configuration.

Do not adjust these settings unless advised to do so by your local dealer or technical support centre. If your system does not appear to be functioning correctly, please contact your local dealer or technical support centre for assistance.

1. Choose **Troubleshooting > Telephone Lines > Audio**. The **Telephone Lines - Audio** window appears.


Telephone Lines - Audio
Telephone Line Audio Calibration
If callers complain that your phone volume is too quiet or too loud, use the Calibrate lines button below. If further adjustment is needed, use the Transmit level menu below to adjust the line volume.
If you encounter echo problems when on calls, some line ports on the system may not be properly calibrated to match the impedance of the lines. Please run the calibration test and if you still have a problem, contact TalkSwitch Support for further instructions.
Calibrate lines...
Unit
Line
Phone Number
Transmit level to callers
1
1
555-1111
Default
Use Defaults
OK
Cancel

2. Calibrate the TalkSwitch unit to match the far end impedance of the telephone lines. Click **Calibrate Lines**. The **Calibrate Lines** window appears.


Calibrate Lines...
Your telephone lines have been previously calibrated but are not optimized.
There are two methods of calibration: OPTIMIZED and QUICK.
The OPTIMIZED method:
requires that your PC be connected on the same LAN as your TalkSwitch unit(s)
provides the best match between your TalkSwitch and your telephone lines
is performed using a separate TalkSwitch Line Optimizer program (included in TalkSwitch software installation)
requires that your TalkSwitch system be out of service while the optimization program is running.
Based on your system's active lines, the OPTIMIZED method will take approximately 4 minutes.
The QUICK method:
provides a good match under typical phone line conditions
can be done without shutting down your telephone system
Based on your system's active lines, the QUICK method will take approximately 2 minutes.
Select one of the calibration methods and click a button below to continue.
QUICK
OPTIMIZED
Proceed
Cancel

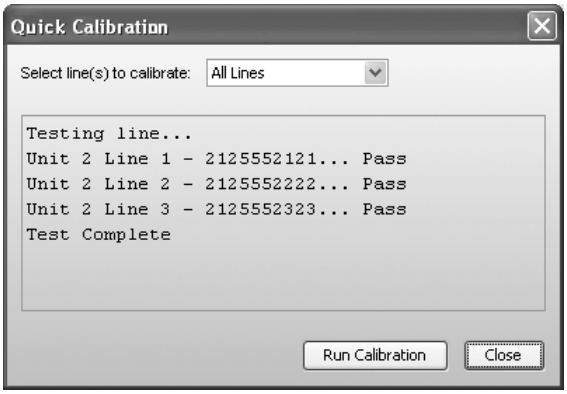
The calibration status of your system is displayed at the top of the window next to the information icon. You may be informed that:

- one or more of your telephone lines requires calibration.
- your telephone lines have been previously calibrated, but are not optimized.
- your telephone lines have been previously optimized and may not require further matching.

3. Select one of the calibration methods by clicking one of the radio buttons at left and click **Proceed** to continue.

QUICK method

a) A **Quick Calibration** window will appear. Use the pull-down menu to select the lines to calibrate, and click **Run Calibration**. The software displays progress information throughout the calibration.



b) When **Test Complete** appears, click **Close**.

OPTIMIZED method

The OPTIMIZED method requires that your PC and your TalkSwitch unit(s) be on the same LAN.

a) You will be prompted to terminate your TalkSwitch configuration session before launching the TalkSwitch Line Optimizer program.

b) The TalkSwitch Line Optimizer should launch upon exit of the management software.

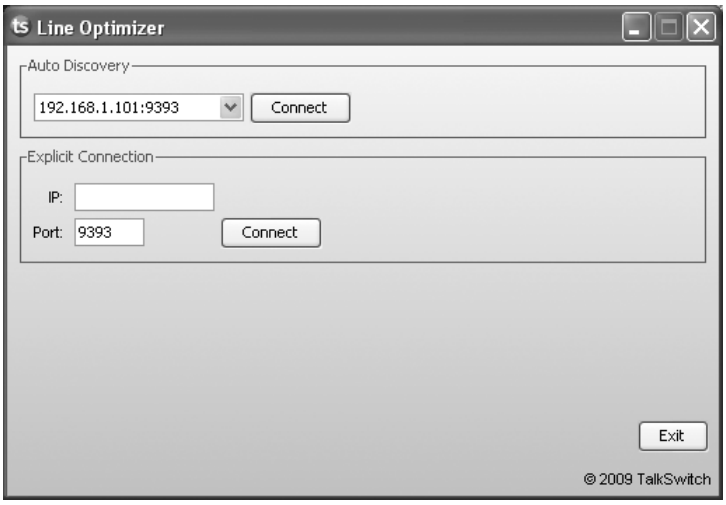
You can also launch this program from the start menu: **Programs > TalkSwitch 6.12 > Line Optimizer**.

If Windows provides a security alert when launching the Line Optimizer program, click **Ignore** or **Unblock** to proceed.

TalkSwitch Line Optimizer

c) The TalkSwitch Line Optimizer program automatically discovers TalkSwitch units on the same LAN that your PC is connected to. Use the pull-down menu to select the unit you want to optimize lines for. Click **Connect**.

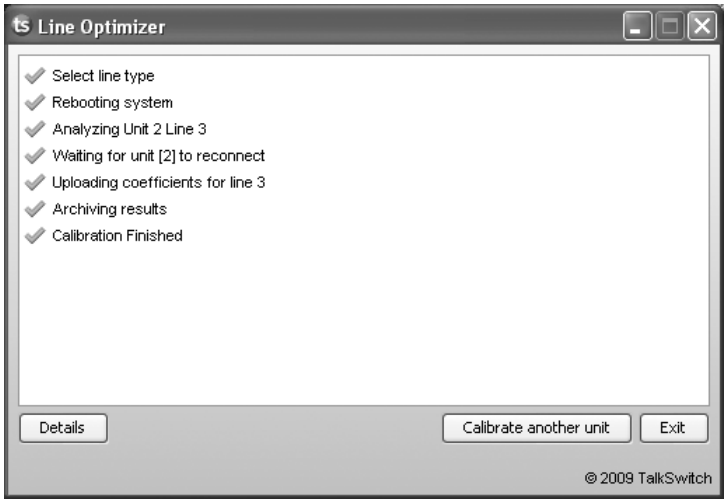
You can also enter the IP address of the desired TalkSwitch unit in the *IP* field in the *Explicit Connection* area. Specify the *Port* if necessary. The default port is *9393*.



d) Use the checkboxes to select the line(s) you want to optimize for each unit, and click **Calibrate...**

e) You will be prompted to allow a reboot of your TalkSwitch system. Click **Yes**.

As your TalkSwitch system reboots and proceeds with optimization, the Line Optimizer displays progress information.



f) When the program reports **Calibration Finished**, you can perform another calibration or exit the Line Optimizer.

4. For each telephone line, set the transmit level to callers. This affects the volume for remote callers, not internal users. Choices are:
   - *Louder*
   - *Default*
   - *Quieter*

CAUTION: Calibrating the telephone lines, and then overriding the outbound volume level can degrade system performance.

Telephone Lines › Detection and Timers

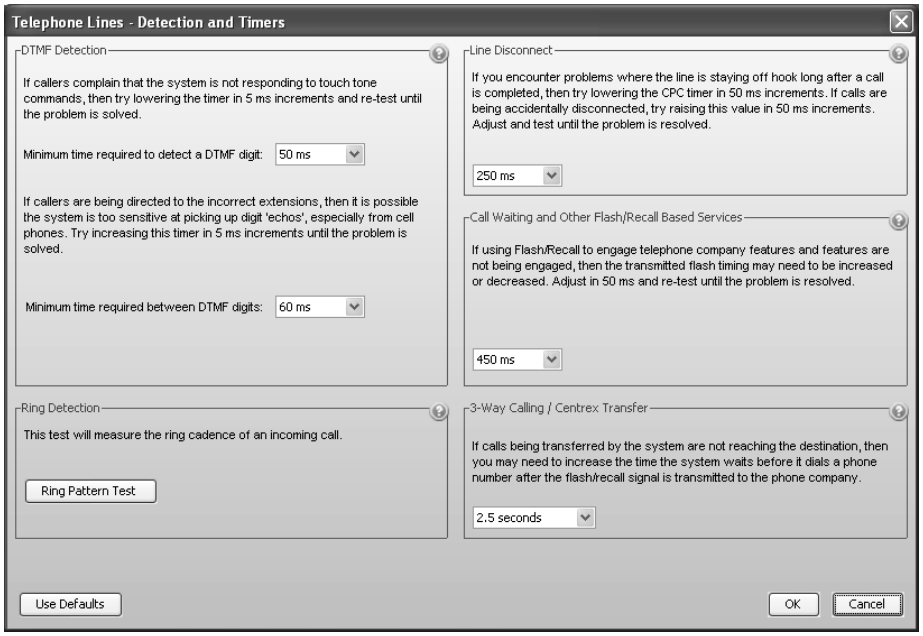
The **Telephone Lines > Detection and Timers** command allows you to set telephone line timers, and perform the ring pattern test.

Do not adjust these settings unless advised to do so by your local dealer or technical support centre. The default settings provide correc operation with telephone lines and analogue extensions.

If your system does not appear to be functioning properly, ensure the region is set correctly (see below), and then click the **Use Defaults** button on this window. If this does not resolve the problem, please contact your local dealer or technical support centre for assistance.

To ensure the region is set correctly, check the country listed in the *Region* field of the **About** page. If it does not match the country where the TalkSwitch unit is operating, change it in the *Region Selection* area of the **Administration** page (if available).

1. Choose **Troubleshooting > Telephone Lines > Detection and Timers**. The **Telephone Lines - Detection and Timers** window appears.


Telephone Lines - Detection and Timers
DTMF Detection
If callers complain that the system is not responding to touch tone commands, then try lowering the timer in 5 ms increments and re-test until the problem is solved.
Minimum time required to detect a DTMF digit:
50 ms
If callers are being directed to the incorrect extensions, then it is possible the system is too sensitive at picking up digit 'echos', especially from cell phones. Try increasing this timer in 5 ms increments until the problem is solved.
Minimum time required between DTMF digits:
60 ms
Line Disconnect
If you encounter problems where the line is staying off hook long after a call is completed, then try lowering the CPC timer in 50 ms increments. If calls are being accidentally disconnected, try raising this value in 50 ms increments. Adjust and test until the problem is resolved.
250 ms
Call Waiting and Other Flash/Recall Based Services
If using Flash/Recall to engage telephone company features and features are not being engaged, then the transmitted flash timing may need to be increased or decreased. Adjust in 50 ms and re-test until the problem is resolved.
450 ms
Ring Detection
This test will measure the ring cadence of an incoming call.
Ring Pattern Test
3-Way Calling / Centrex Transfer
If calls being transferred by the system are not reaching the destination, then you may need to increase the time the system waits before it dials a phone number after the flash/recall signal is transmitted to the phone company.
2.5 seconds
Use Defaults
OK
Cancel

2. Select the *Minimum required time to detect a DTMF digit*. This is how long the caller must press the number key on their phone for the TalkSwitch unit to detect a valid digit. Choices range from *40 ms.* to *100 ms*.

   Adjust this if callers are misdirected after selecting options using the auto attendant. Use a shorter time if the TalkSwitch unit appears to have trouble hearing the digits. Use a longer time if the auto attendant is misdirecting callers.

3. Select the *Minimum required time between DTMF digits*. This is how long the caller must wait after releasing the number key on their phone for the TalkSwitch unit to detect the next digit. Choices range from *30 ms.* to *80 ms*.

Use a longer time if the auto attendant is misdirecting callers due to echoed digits from cell phones. For example, calls are misdirected to extension 111. This can happen if the caller dials a different extension number (e.g. 113), but the telephone network echoes the 1, so the TalkSwitch unit hears 1113. It captures the first three digits, ignores the 3, and misdirects the call to extension 111.

4. Select the *Line disconnect time*. This is how long the CPC signal must last for the TalkSwitch system to hang up the telephone line. The telephone company sends the CPC or Disconnect Clear signal when the caller hangs up. Choices range from *100 ms.* to *600 ms*.

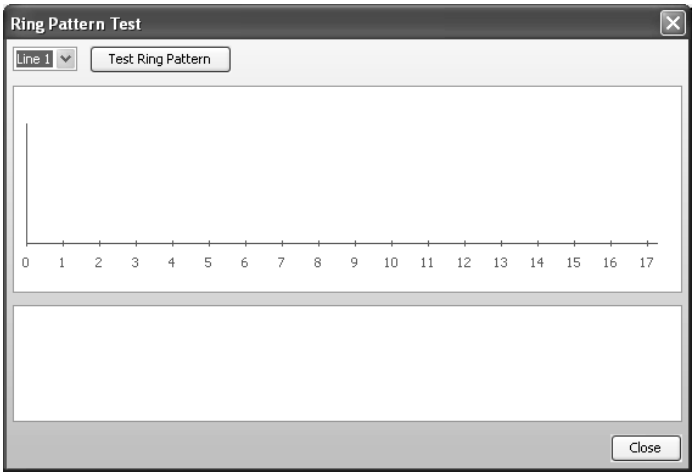
   Use a longer time if calls are accidentally disconnected during a conversation. This can occur if there is a problem with the telephone line. If the line has power fluctuations, or the line card at the central office is defective, spurious CPC signals can occur that cause the TalkSwitch unit to hang up.

5. Set the *Transmitted flash length* to the duration of the Flash sent by the TalkSwitch system to the telephone company in order to control telephone company services. Choices range from *250 ms.* to *1000 ms*.

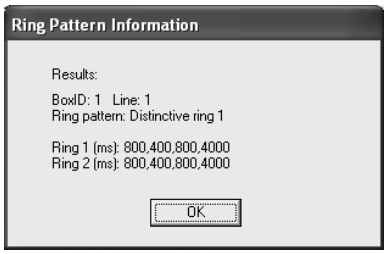
6. Set the *3-way calling wait time* to the minimum interval between dialing the Flash signal and the telephone number. The 3-Way Calling/Conference service requires this interval to re-establish dial tone. Choices range from *500 ms.* to *5 seconds*.

7. To test the ring pattern for incoming calls on telephone lines:

   a) Click **Ring Pattern Test**. The **Ring Pattern Test** window appears.



   b) Click **Test Ring Pattern**. Call one of the telephone lines. After three rings the **Ring Pattern Information** window appears. The time is displayed in milliseconds.



VoIP

VoIP ports

The SIP server port and starting RTP port can be specified for the SIP server for a TalkSwitch VoIP network. See *VoIP ports* on page 188.

SIP Server Registration Timer

The SIP Server Registration Timer can be specified for the SIP server for a TalkSwitch VoIP network. See *SIP server registration* on page 188.

UPnP

By default, TalkSwitch uses Universal Plug'n Play (UPnP) to attempt to automatically open the required ports for VoIP communication through supported firewalls. By disabling this feature, the ports must be manually configured within your firewall in order for VoIP communication to operate.

For information on configuring your router ports for TalkSwitch VoIP networks, VoIP service providers and external IP extensions, see *Router port forwarding* on page 36.

View Blocked IP Addresses

If TalkSwitch detects IP traffic that appears malicious or may impair operation of your phone system, it will block traffic from the source IP address. The *View Blocked IP Addresses* area provides a hyperlink to a page that displays blocked IP addresses.

Test Firewall

The **Troubleshooting > Test Firewall** command allows you to test the status of each port through the firewall. See *Router port forwarding* on page 36.

1. Choose **Troubleshooting > Test Firewall**. The **Test Firewall** window appears.

2. Select the services you want to check.
3. Click **Test Ports**. The system will check the ports of the selected services.

Support

The **Troubleshooting > Support** command allows you to automatically e-mail log files to TalkSwitch Customer Support. Only use this command as directed, after receiving a Case ID from TalkSwitch Customer Support.

Automatically e-mailing log files requires Microsoft Outlook to be configured as your default e-mail client. Note that the e-mail will only include information related to your TalkSwitch system.

1. Choose **Troubleshooting > Support**. The **Support** window appears.

Support
TalkSwitch Management Software will capture your current system information and generate an email to TalkSwitch Support.
Note: No personal information is captured in the file.
Case ID provided by TalkSwitch Support:
To use an email client other than your default client, please send an email to support@talkswitch.com with a description of the case scenario and attach the file at the link below:
es\TalkSwitch\TalkSwitch Configuration 6.10\Logs\TSPackage__6.10005_1207249240773.zip
OK
Cancel

2. Enter the *Case ID* supplied by TalkSwitch Customer Support.

If you use a different e-mail client, or don't have one configured (e.g. you use a web-based e-mail service like Hotmail) you can manually e-mail the log files. Address the e-mail to support@talkswitch.com. Enter the Case ID in the Subject line. Attach the .zip file listed in the **Support** window.

CHAPTER 3

USING TALKSWITCH

RECEIVING CALLS

With the auto attendant

If you don't have a receptionist, the auto attendant helps you handle all your inbound calls. Once an auto attendant has been configured and recorded, it will follow the instructions given in the configuration.

When calls are answered by an auto attendant, callers can:

- Dial any local or remote extension
- Dial any VoIP number (250–299)
- Dial any ring group (300–309)
- Dial voicemail directly
- Use the dial-by-name directory
- Choose options **0** (**9** in some regions), **1**, **2**, **3**, **4**, **5** (if configured to perform specific actions)
- Transmit a fax
- Go to a different auto attendant
- Remain on the line to follow an action programmed for that auto attendant

TalkSwitch users can access other options at the auto attendant:

- Call back or call bridge
- Retrieve voicemail
- Enter command mode to make system changes

All these features can be password-protected to avoid unauthorized access.

TalkSwitch also provides the option of ringing specific extensions or ring groups prior to the engagement of an auto attendant.

Without the auto attendant

If you do not use the auto attendant to answer calls, please note that the call cascade options for the local extensions are not engaged. For calls to go to an extension's mailbox, the call must be transferred manually by someone in the office. Once calls have been answered at a local extension, they can be transferred to any extension or voice mailbox.

USING AN ANALOG EXTENSION

This section describes how to use a third party analog phone with the TalkSwitch system. You can also use these procedures with the TalkSwitch TS-80, TS-200, TS-400, and TS-600 phones. However, these phones have additional keys that make it easier to access the functions. Refer to the user guide supplied with your TalkSwitch phone.

Making calls

Placing an intercom call

An intercom call is a call from a local extension to another local extension, remote extension or ring group.

1. Dial the extension number.

If the *Intercom calls follow cascade settings* check box is selected in the configuration, an intercom call will follow the call cascade of the dialed extension. Otherwise the intercom call will ring continuously.

The dialed extension rings as follows:

- A local extension uses two short rings to identify the intercom call.
- A remote extension rings normally.
- A ring group uses the ring pattern selected in the configuration. The ring group uses the same ring pattern for intercom calls and outside calls.

Overhead paging

Overhead paging is a call from a local extension through the optional PA amplifier connected to the TalkSwitch unit.

1. Dial ***0**.

Two-way paging

Two-way paging is a call from a local extension to another local extension that activates the speakerphone of the dialed extension. The user at the dialed extension can reply through the speakerphone. You can perform two-way paging from a local extension to a TalkSwitch analog phone (TS-200, TS-400 or TS-600), or most TalkSwitch IP phone models (TS-9112i, TS-9133i, TS-350i, TS-480i or TS-550i).

1. Dial ***84**.
2. Dial the extension number.

One-way group paging

One-way group paging is a call from a local extension to another local extension or to a ring group that activates the speakerphones of the dialed extensions. The users at the dialed extensions cannot reply through the speakerphone. You can perform one-way group paging from a local extension to a TalkSwitch analog phone (TS-200, TS-400 or TS-600).

1. Dial ***85**.
2. Dial the extension or ring group number.

Place out-of-office calls from a local extension

1. Dial **9** (**0** in some regions) or **81**–**88** to choose a hunt group. The hunt group gives you access to an available line to dial out.
2. If your system requires permissions for outgoing calls, enter your access code.
3. Dial the phone number.

Extensions can be restricted to specific hunt groups, and can block certain outbound calls. See *Hunt group permissions* on page 64 and *Auto Route Selection* on page 160.

Your extension can also be configured with direct line access. When you pick up the handset, you hear the telephone company dial tone and can dial an outside number without dialing a hunt group number. See *Direct line access* on page 61.

If direct line access is enabled for a local extension, you need to press **Flash** or **Recall** before dialing any extensions or voice mailboxes.

If you dial your own extension you will be directed to your voicemail. If your voicemail is not set up, you will get a busy signal.

Receiving calls

The steps below apply when an analog extension is ringing. If an auto attendant engages before you are able to pick up, the caller can select an option at the auto attendant.

For your local extension

You can answer a call that is ringing your local extension.

1. If the call went through the auto attendant and call screening is enabled:
   - Press **#** to accept the call.
   - Press ***** to reject and re-route the call to its extension's call cascade, and then hang up.

For any other local extension

You can intercept a call that is ringing another local extension.

1. Press ***9**.

For a specific local extension

You can intercept a call that is ringing a specific extension.

1. Press ***7**.
2. Dial the extension number.

While you're on a call

You can intercept a call while connected to another caller.

1. Press **Flash** or **Recall** to place the first call on hold.
2. Press ***9,** or ***7** and the extension number to intercept the second call.
3. Once you are finished with the second call or would like to return to the first caller, press **Flash 5** or **Recall 5**.

Placing calls on hold

1. Press **Flash** or **Recall** to place the call on hold.
2. Press **Flash** or **Recall** again to take the call off hold.
3. If you have multiple callers on hold or in queue at your extension, press **Flash 7** or **Recall 7** to retrieve callers on a first in/first out basis.

If you put a caller on hold using the **Hold** button on some third party analog phones, you won't be able to access Talkswitch features while the caller is holding. If you have music on hold enabled, the caller will not hear music while on hold.

While a call is on hold, your phone will ring once each time the reminder timer expires. See *Call Reminders on page 187*.

Transferring calls to an extension

You can initiate a transfer from a local extension to another local extension, remote extension or ring group.

Unscreened transfer

1. Press **Flash** or **Recall** to put the caller on hold.
2. Dial the extension you want to transfer to, and then hang up. The caller is taken off hold when the receiving extension answers. If the extension is busy or the call not answered, the system can either perform the call cascade's Busy action or No Answer action, or can ring back the extension that transferred the call. See *To local extension tab* on page 184.

Screened transfer

1. Press **Flash** or **Recall** to put the caller on hold.
2. Dial the extension number you want to transfer to, and wait for them to answer.
    - If the person at the receiving extension wishes to take the call, complete the transfer by hanging up or pressing **Flash 4** or **Recall 4**. The caller is immediately taken off hold and connected to the receiving extension. You hear an internal dial tone.
    - If the person at the receiving extension does not wish to take the call, return to the caller by pressing **Flash 5** or **Recall 5**.
    - If you dial an extension that is busy or not answered, you can complete the transfer by hanging up or pressing **Flash 4** or **Recall 4**. The call will be routed according to that extension's call cascade settings. You hear an internal dial tone.

If you handle many calls, you may want to complete or cancel a transfer without hanging up. **Flash 4** or **Recall 4** will complete the transfer and provide an internal dial tone. **Flash 5** or **Recall 5** cancels the transfer and connects you back to the caller.

Transferring calls to an outside number

You can initiate a transfer from a local extension to an outside number.

If the telephone line has the 3-Way Calling/Conference service or the Transfer and Clear service, and the local extension has enabled *Allow the use of same line connect (by dialing 80)*, the user can select hunt group 80 to dial the outside number on the same line the caller is on.

Unscreened transfer

1. Press **Flash** or **Recall** to put the caller on hold.
2. Dial **9** (**0** in some regions) or **81**–**88** to choose a hunt group.
3. Dial the outside phone number.
4. Press **Flash 4** or **Recall 4** to complete the transfer. Do not hang up to complete the transfer, because this disconnects the call.

Screened transfer

Instead of completing the transfer once you have dialed the outside number, wait until the call is answered and ask the recipient if he or she wishes to take the call. If the recipient accepts the call, press **Flash 4** or **Recall 4** to complete the transfer. If not, press **Flash 5** or **Recall 5** to return to the caller.

Parking and retrieving calls

Call park is a feature for placing a call on hold and then retrieving it from another local extension. TalkSwitch has 10 park orbits (500–509).

Parking a call

1. Press **Flash** or **Recall** to put the caller on hold.
2. Press ***510**. TalkSwitch selects the first available park orbit (500–509). If you use a TS-80, TS-200, TS-400 or TS-600 phone, press the **Park** button or softkey. You hear a confirmation indicating the caller has been parked successfully and into which park orbit.
3. Provide the park orbit to the person with the parked call (e.g. *"Mary, there is a call parked for you in 500"*).

While a call is parked, your phone will ring once each time the reminder timer expires. See *Call Reminders on page 187*.

Retrieving a parked call

1. If direct line access is enabled at your extension, press **Flash** or **Recall** to get an internal dial tone.
2. Press ** and the appropriate park orbit. If you use TS-80, TS-200, TS-400 or TS-600 phone, press the **Un-park** button or softkey and enter the park orbit.

Queuing and retrieving callers

Call queue is a useful feature for placing multiple callers on hold at your extension while you are on an existing call. For details on configuring call queue, please refer to *How call cascades work* on page 65.

Queuing callers at a local extension

If an extension is busy and the *Busy* call cascade is set to *queue at extension*, incoming callers hear one of the prompts below, followed by music (if enabled) while they are on hold:

- If the call came from the auto attendant and the extension has a voice mailbox: *"The extension you have reached is currently busy. If you wish to continue holding, please remain on the line. To leave a voicemail message, press 1, and to return to the main menu, press star."*

- If the call came from the auto attendant and the extension has no voice mailbox: *"The extension you have reached is currently busy. If you wish to continue holding, please remain on the line. To return to the main menu, press star."*
- If the call was transferred from another extension and the extension has a voice mailbox, but there is no auto attendant on this line with a main menu to return to: *"The extension you have reached is currently busy. If you wish to continue holding, please remain on the line. To leave a voicemail message, press one."*
- If the call was transferred from another extension and the extension has no voice mailbox and there is no auto attendant on this line with a main menu to return to: *"The extension you have reached is currently busy. If you wish to continue holding, please remain on the line."*

While the caller is on hold at your extension, you hear a call waiting beep every 75 seconds. If you want to rotate through queued callers, press **Flash 7** or **Recall 7**. The caller you are speaking to is placed on hold, and the first caller in the queue is retrieved. If you have multiple callers queued, press **Flash 7** or **Recall 7** to retrieve callers on a first in/first out basis.

If an extension is busy and the *Busy* call cascade is set to *invoke call waiting*, callers hear ringing (or music if enabled) while the local extension hears a call waiting beep and receives the caller ID for the new call. To connect to the new caller, press **Flash 7** or **Recall 7**. To let the caller fall through to the next level of the cascade, ignore the beep.

Queuing callers at a ring group

If you want to queue callers at a ring group, configure this action using an auto attendant.

For example, configure auto attendant 1 to queue callers to ring group 301 for technical support if they have pressed **5**. When a caller presses **5** for technical support, they enter the queue immediately.

Every 60 seconds (configurable) the caller hears the following prompt: *"Your call is important to us. Please remain on the line. To return to the previous menu, press star."* Extensions that are part of the ring group and are available will start ringing within 10 seconds of hanging up the phone from a previous call. The delay allows time to make a new call or activate *Do not Disturb* as a log out option. After 5 extension rings, available phones will ring once every 16 seconds. When you pick up, you hear a prompt that provides the option to accept the queued call by pressing **#** or leave the caller in the queue by pressing ***** and returning to the dial tone to make a new call.

Using call waiting

The call waiting feature is activated by setting the *If extension is busy* list to *invoke call waiting*. This control is in the *Busy* tab of the **Local Extensions/Fax** page. When the dialed local extension is busy, the caller hears ringing or music depending on the settings.

If the phone supports call waiting caller ID (type II signaling), you hear a beep followed by a short blip. Depending on your phone, you may hear something similar to a modem noise. This is normal. During this notification, the caller ID information is displayed on the phone. The caller does not hear the beep. The phone mutes the other side.

If the phone does not support call waiting and caller ID, you hear a beep. When the call waiting notification beep is heard, you can either press **Flash 7** or **Recall 7** to put the current caller on hold and answer the new call, or press **Flash 5** or **Recall 5** to terminate the current call and answer the new call. If you do not want to answer the new call, ignore the call waiting beeps. TalkSwitch will direct the caller to the next level of the call cascade for your extension (e.g. TalkSwitch will send the caller to your voice mailbox).

Conference calling

You can set up conference calling between three local extensions, two local extensions and one outside caller, or two outside callers and one local extension.

Two local TalkSwitch extensions and one outside caller

You do not require access to the telephone company's 3-Way Calling/Conference service to use the TalkSwitch conference calling capabilities.

1. Establish a call with an outside party.
2. Press **Flash** or **Recall** to place the outside caller on hold.
3. Dial the number of the local extension you wish to conference with.
4. When the extension is picked up, press **Flash 6** or **Recall 6** to establish the 3-way call.

Two outside callers and one local extension

There are two different methods for this type of conference call. The first method is similar to the one above, using only TalkSwitch for the conference function.

1. Establish a call with an outside party.
2. Place the outside caller on hold.
3. Dial **9** (**0** in some regions) or **81**–**88** to choose a hunt group.
4. Dial the number of the other outside party.
5. When the call is answered, press **Flash 6** or **Recall 6** to establish the 3-way call.

The second method requires the same line connect feature and uses the telephone company's 3-Way Calling/Conference service.

1. Establish a call with an outside party.
2. Place the outside caller on hold.
3. Dial **80** to choose same line connect.
4. Dial the number of the other outside party.
5. When the call is answered, press **Flash 6** or **Recall 6** to establish the 3-way call.

The conference initiator can disengage the second conferenced party at any time by pressing **Flash 5** or **Recall 5**.

USING AN IP EXTENSION

This section describes how to use a third party IP phone with the TalkSwitch system. You can also use these procedures with the TalkSwitch IP phones. However, these phones have additional keys that make it easier to access the functions. See *TalkSwitch phone programmable key functions* on page 214, and refer to the guide supplied with your TalkSwitch phone.

Making calls

Placing an intercom call

An intercom call is a call from a local extension to another local extension, remote extension or ring group.

1. Dial the extension number.

If the *Intercom calls follow cascade settings* check box is selected in the configuration, an intercom call will follow the call cascade of the dialed extension. Otherwise the intercom call will ring continuously.

The dialed extension rings as follows:

- A local extension uses two short rings to identify the intercom call. IP phones supported by TalkSwitch can identify intercom calls this way.
- A remote extension rings normally.
- A ring group uses the ring pattern selected in the configuration. The ring group uses the same ring pattern for intercom calls and outside calls.

Overhead paging

Overhead paging is a call from a local extension through the optional PA amplifier connected to the TalkSwitch unit.

1. Dial ***0**.

Two-way paging

Two-way paging is a call from a local extension to another local extension that activates the speakerphone of the dialed extension. The user at the dialed extension can reply through the speakerphone. You can perform two-way paging from a local extension to a TalkSwitch analog phone (TS-200, TS-400 or TS-600), or most TalkSwitch IP phone models (TS-9112i, TS-9133i, TS-350i, TS480i or TS-550i).

1. Dial ***84**.
2. Dial the extension number.

One-way group paging

One-way group paging is a call from a local extension to another local extension or to a ring group that activates the speakerphones of the dialed extensions. The users at the dialed extensions cannot reply through the speakerphone. You can perform one-way group paging from a local extension to a TalkSwitch analog phone (TS-200, TS-400 or TS-600).

1. Dial ***85**.
2. Dial the extension or ring group number.

Place out-of-office calls from a local extension

Warning: Using an external IP extension to call an emergency service number will not send the correct address to the emergency operator. We strongly recommend that you apply a warning label to any external IP extension stating:

If an emergency call is made from this phone, you must provide your address to the emergency operator.

1. Dial **9** (**0** in some regions) or **81**–**88** to choose a hunt group. The hunt group gives you access to an available line to dial out.
2. If your system requires permissions for outgoing calls, enter your access code.
3. Dial the phone number.
4. Depending on your IP phone, press **Dial**, **Send** or **#**.

Extensions can be restricted to specific hunt groups, and can block certain outbound calls. See *Hunt group permissions* on page 64 and *Auto Route Selection* on page 160.

Your extension can also be configured with direct line access. When you pick up the handset, you hear the telephone company dial tone and can dial an outside number without dialing a hunt group number. See *Direct line access* on page 61.

If direct line access is enabled for a local IP extension, you need to press ** before dialing any extensions or voice mailboxes.

Receiving calls

If an auto attendant engages before you are able to pick up, the caller can select an option at the auto attendant. The steps below apply when you pick up an extension that is ringing.

For your local extension

You can answer a call that is ringing your local extension.

1. If the call went through the auto attendant and call screening is enabled:
   - Press **#**, then press **Dial, Send** or **#** to accept the call.
   - Press *, then press **Dial, Send** or **#** to reject and re-route the call to its extension's call cascade.

For any other local extension

You can intercept a call that is ringing on any other local extension.

1. Press ***9**, then press **Dial, Send** or **#**.

For a specific local extension

You can answer a call that is ringing at a specific extension.

1. Press ***7**.
2. Dial the extension, then press **Dial, Send** or **#**.

While you're on a call

You can intercept a call while connected to another caller.

1. Press **Hold** to place the first call on hold.
2. Press ***9**, or ***7** and the extension number, then press **Dial, Send** or **#**.
3. Once you are finished with the second call or would like to return to the first caller, press **Cancel**.

Placing calls on hold

1. Press **Hold**.
2. If you have multiple callers on hold or in queue at your extension, you can retrieve them on a first in/first out basis. Depending on your phone, you can either:
   - Press the flashing line key that corresponds to the caller on hold.
   - Press **Hold** and **7**, then press **Dial, Send** or **#**.

Transferring calls to an extension

Unscreened transfer

1. Press **Transfer** or **xfer** to put the caller on hold.
2. Dial the extension number.
3. Press **Transfer** or **xfer**. If the IP phone supports the hang-up transfer feature, hang up the phone. Please check the documentation of the IP phone to see if it supports this feature.

The caller is taken off hold when the receiving extension answers. If the extension is busy or the call not answered, the system can perform the call cascade's Busy action or No Answer action, or can ring back the extension that transferred the call. See *Transfer Preferences* on page 184.

Screened transfer

To initiate a screened transfer from an IP extension to another local extension, remote extension or ring group:

1. Press **Transfer** or **xfer** to put the caller on hold.
2. Dial the extension number.
   - If the person at the receiving extension wishes to take the call, press **Transfer** or **xfer** to complete the transfer. If the IP phone supports the hang-up transfer feature, hang up the phone. Please check the documentation of the IP phone to see if it supports this feature. The caller is immediately taken off hold and connected to the receiving extension.
   - If the person at the receiving extension does not wish to take the call and you want to return to the caller, press **Cancel**. If this softkey is unavailable, press the flashing line button that corresponds to the caller on hold. You can also press **Hold** and **7**, then press **Dial, Send** or **#**.
   - If you dial an extension that is busy or not answered, you can complete the transfer. The caller will be directed according to that extension's call cascade settings.

Transferring calls to an outside number

You can initiate a transfer from an IP extension to an outside number.

If the telephone line has the 3-Way Calling/Conference service or the Transfer and Clear service, and the local extension has enabled *Allow the use of same line connect (by dialing 80)*, the user can select hunt group 80 to dial the outside number on the same line the caller is on.

Unscreened transfer

1. Press **Transfer** or **xfer** to put the caller on hold.
2. Dial **9** (**0** in some regions) or **81**–**88** to choose a hunt group.
3. Dial the outside phone number.
4. Press **Transfer** or **xfer** to complete the transfer. Do not hang up to complete the transfer, because this disconnects the call.

Screened transfer

1. Press **Transfer** or **xfer** to put the caller on hold.
2. Dial **9** (**0** in some regions) or **81**–**88** to choose a hunt group.
3. Dial the outside phone number.
4. Wait until the call is answered and ask the recipient if he or she wishes to take the call. If the recipient accepts the call, press **Transfer** or **xfer** to complete the transfer. If not, press **Cancel** to return to the caller. If this softkey is unavailable, press the flashing line button that corresponds to the caller on hold.

Call park — parking and retrieving callers

Call park is a feature for placing a call on hold and then retrieving it from any other local extension. TalkSwitch has 10 park orbits, 500–509.

Parking a caller

1. Press **Transfer** or **xfer** to put the caller on hold.
2. Press ***510**, then press **Dial**, **Send** or **#**. TalkSwitch selects the first available park orbit (500–509). You hear a confirmation indicating the caller has been parked successfully and into which park orbit.
3. Provide the park orbit to the person with the parked call (e.g. *"Mary, there is a call parked for you in 500"*).

Retrieving a parked call

1. Press ******, the park orbit, then press **Dial**, **Send** or **#**.

Queuing and retrieving callers

This feature is used the same way on analog and IP extensions except for the slight differences outlined in the next paragraph.

Some IP extensions display a notification on the screen while the caller is on hold at your extension. If you want to rotate through queued callers while you are on the phone, dial **7** and press **Dial, Send** or **#**. The caller you are speaking to is placed on hold, and the first caller in the queue is retrieved. If several callers are queued, dial **7** and press **Dial, Send** or **#** to retrieve them on a first in/first out basis.

Using call waiting

If the call waiting feature is activated as a *Busy* option in the call cascade routing for a local extension and the dialed extension is in use, the caller hears ringing or music depending on the music on hold settings. You hear the following:

- If the phone supports call waiting caller ID, you hear a beep followed by a short blip.
- If the phone does not support call waiting caller ID, you hear a beep. When the call waiting notification beep is heard, you can pick up another line and dial **7** and press **Dial, Send** or **#** to put the current caller on hold and answer the new caller. To return to the previous call, press the flashing line key that corresponds to the previous call. On most IP extensions, this will put the current call on hold and connect to the other caller.
- To terminate the current call and go back to the other caller on hold, hang up, then go off-hook and press the button that corresponds to the flashing line of the caller on hold. If you do not want to take the new call, ignore the call waiting beeps. TalkSwitch will direct the caller to the next level of the call cascade for that extension (i.e. send the caller to the associated voice mailbox).

Conference calling with TalkSwitch

Note: If you try to initiate a conference call with a TalkSwitch 9112i IP phone, but there is no line available, you will not hear a busy signal. You will instead be reconnected with the other caller.

Two TalkSwitch local extensions and one outside caller

You do not require access to the telephone company's 3-Way Calling/Conference service to use the TalkSwitch conference calling capabilities.

1. Establish a call with an outside party.
2. Press **Conf** or **Hold** to place the outside caller on hold.
3. Dial the local extension you wish to conference with.
4. When the extension answers, press **Conf** to establish the 3-way call.

Two outside callers and one local extension

There are two different methods for this type of conference call. The first method is similar to the one above, using only TalkSwitch for the conference function.

1. Establish a call with an outside party.
2. Press **Conf** or **Hold** to place the outside caller on hold.
3. Dial **9** (**0** in some regions) or **81**–**88** to choose a hunt group.
4. Dial the number of the other outside party.
5. When the call is answered, press **Conf** to establish the 3-way call.

The second method uses the same line connect feature and the telephone company's 3-Way Calling/Conference service.

1. Establish a call with an outside party.
2. Press **Hold** to place the outside caller on hold.
3. Dial **80** to choose the hunt group.
4. Dial the number of the other outside party and press **Dial, Send** or **#**.
5. When the call is answered, press **Transfer** to establish the 3-way call.

TALKSWITCH PHONE PROGRAMMABLE KEY FUNCTIONS

Many TalkSwitch IP phone models offer programmable keys. The function and associated resources are assigned using TalkSwitch Management Software in the **Local Extensions/Fax > IP Extension Details > Configure Keys** page. Supported functions for a phone model typically include most of the items listed below.

- *Line appearance* — Select a telephone line or VoIP number as the resource. The corresponding button or softkey will display the status of the line, and allow you to make calls with a single press of the button or softkey.

  For phones equipped with button lights, the button will light up when the line is in use, flash if the line is ringing, or be off when the line is available.

  For phones with softkeys, a status icon and line ID will be displayed beside the softkey. The display will show an off-hook icon when the line is in use, show a ringing icon when the line is ringing, and show an on-hook icon when the line is available.

- *Extension appearance* — Select a local extension as the resource.

  The corresponding button or softkey will display the status of the selected extension, and allow you to call the extension with a single press of the button or softkey.

  For phones equipped with button lights, the button will light up when the selected extension is in use, flash if the extension is ringing, or be off when the extension is available.

  For phones with softkeys, a status icon and extension call ID will be displayed beside the softkey. The display will show an off-hook icon when the selected extension is in use, show a ringing icon when the extension is ringing, and show an on-hook icon when the extension is available. The extension call ID is also be displayed.

- *Queue appearance* — Select a local extension as the resource. The corresponding button or softkey will indicate whether calls are queued at the selected extension, and allow you to pick up the oldest queued call with a single press of the button or softkey.

  For phones equipped with button lights, the button will flash if calls are queued for the extension, or be off when there are no queued calls.

  For phones with softkeys, a status icon, extension number and "Q" will be displayed beside the softkey. The display will show a ringing icon if calls are queued for the extension, or an on-hook icon when there are no queued calls.

- *Voicemail* — Do not select a resource. Press the button or softkey to access the voice mailbox of the local extension. Lights or icons are not used for voicemail keys. Note: You can also access Voicemail by pressing ****#**.

- *DND* — Do not select a resource. Press the button or softkey to toggle Do Not Disturb mode on or off. Lights or icons are not used for DND keys. Note: You can also toggle DND mode by pressing ***62#**.
- *Park* — Do not select a resource. Press the button or softkey to put the call on hold, in the next available park orbit. The system will respond with the park orbit number (500 to 509). Lights or icons are not used for Park keys. Note: You can also Park a call by pressing /***510#** on a TS-350i, or ***510#** on a TS-550i, or Xfer ***510#** on a TS-9112i, TS-9133i or TS-480i.
- *Un-park* — Do not select a resource. Press the button or softkey, select the park orbit number (500 to 509), then press Unpark to retrieve the call. Softkey displays will show the parked call ID beside the orbit number. Lights or icons are not used for Unpark keys. Note: You can also Unpark a call by lifting the handset, pressing *****, dialing the park orbit number, then pressing **#**.
- *Pickup any* — Do not select a resource. Press the button or softkey to answer a call from an outside number ringing any extension. Lights or icons are not used for Pickup keys. Note: You can also pick up a call by pressing ***9#**.
- *Pickup ext* — Do not select a resource. Press the button or softkey, dial an extension, then dial **#** (or softkey again) to answer a call from an outside number or extension ringing the selected extension. Lights or icons are not used for Pickup keys. Note: You can also pick up a call at a selected extension by pressing ***7**, dialing the extension, then pressing **#**.
- *Intercom page* — Do not select a resource. Press the button or softkey, dial an extension, then press **#** to page the extension in Intercom mode. The intercom page function can only page TalkSwitch phones. Lights or icons are not used for Page keys. Note: You can also page a TalkSwitch phone by pressing ***84**, dialing the extension, then pressing **#**, or the **Dial** key or softkey.
- *Overhead page* — Do not select a resource. Press the button or softkey to connect to an attached external PA system. Lights or icons are not used for Page keys. Note: You can also page the external PA system by pressing ***0#**.
- *Queue (show list)* — Do not select a resource. Press the **Queue** softkey, select a queued call, then press **Retrieve** to connect to the selected call. This function is only available when programmed as a function key on supported phone models (e.g. TS-480i).
- *Phone book access* — Select the phone book record two-digit speed dial number as the resource. Press the button or softkey to place a call using the contact information from the associated phone book record. When the button is pressed, it will light up for the duration of the call. This function is available on the TS-350i and TS-550i models.
- *User Defined (phone)* — The button or softkey function is assigned using the phone's configuration options, and is not assigned using TalkSwitch Management software.
- *None* — The button or softkey is not programmed.

USING VOIP

A TalkSwitch VoIP network provides inter-branch connectivity. It requires a VoIP-enabled TalkSwitch unit at each location. A TalkSwitch VoIP network has 50 VoIP numbers (250–299) that can be dialed from an extension or auto attendant. The TalkSwitch unit automatically finds the VoIP lines for placing these calls.

Other VoIP calls are made by dialing a hunt group. If a VoIP-enabled TalkSwitch unit is present, hunt group 88 provides access to the TalkSwitch VoIP network by default. Refer to the *VoIP Network Configuration Guide*. It tells you how to set up a TalkSwitch profile or service provider profile, set up VoIP numbers that use the profile, and set up the hunt group. Dialing the hunt group at the internal dial tone tells the TalkSwitch system to hunt for an available VoIP line.

A VoIP-enabled TalkSwitch unit is required for teleworkers to use external IP extensions. The external IP extensions are local extensions, and function like internal IP extensions at the office.

Example: You have three branch locations (San Francisco, New York, Miami).

Location 1: San Francisco
Twelve VoIP numbers (250–261) are assigned to four VoIP lines.

Location 2: New York
Four VoIP numbers (270–273) are assigned to four VoIP lines.

Location 3: Miami
Four VoIP numbers (280–283) are assigned to four VoIP lines.

If someone in the San Francisco office wants to reach someone in the New York office, they can dial one of the VoIP numbers associated with New York (270–273). TalkSwitch automatically finds an available VoIP line and connects to the New York location. The TalkSwitch unit at the New York location handles the call according to the configuration in the **VoIP Numbers** page.

USING A STANDARD PHONE CONNECTED IN PARALLEL TO TALKSWITCH

A standard phone connected in parallel (not connected to TalkSwitch but connected to the same line) with TalkSwitch always rings at least once before TalkSwitch starts ringing its extensions or the auto attendant answers the call. TalkSwitch uses the first ring to identify the incoming ring pattern before passing it on to the extensions or the auto attendant (TalkSwitch determines if the call is a distinctive ring and detects any caller ID information). A parallel phone can interrupt a telephone call on any of the telephone lines.

Making and receiving calls

Make calls on parallel-connected extensions as if TalkSwitch were not sharing the telephone line with them.

If you answer on a parallel-connected phone before the auto attendant engages, TalkSwitch takes no further action. If a call is answered on a parallel-connected phone after the auto attendant engages, you can turn off the auto attendant message by pressing ****0** at one of the TalkSwitch extensions, or let it time-out on its own, in which case you will hear: *"Good-bye"*.

Transferring calls

Calls answered on a parallel-connected phone can be transferred to any TalkSwitch extension or voice mailbox. To transfer the caller, press ******. After you hear 2 beeps, dial the extension and hang up. You cannot announce a caller to the transferred extension but if you stay on the line, you are in a conference call state. You can hang up at any time.

If you transfer a call from a parallel-connected phone and then hang up, the extension rings the number of times specified for that extension. If it is not answered within that number of rings, the call is handled according to that extension's call cascade actions.

When a phone in parallel with TalkSwitch is off-hook, you notice that the corresponding line light flickers to indicate the line is busy. TalkSwitch does not allow extensions to interfere on a call that has the line in use.

You can change the transfer prefix to ***#** if the default transfer prefix conflicts with another device or service. See *From home phone tab* on page 187.

MODEMS AND TELEPHONE LINE ACCESS

If your modem is attached to TalkSwitch as an extension, you can configure it to access your telephone lines in one of two ways.

1. Enable direct line access to one of your telephone lines. You don't have to dial **9** (**0** in some regions) or any other hunt group to access an outside line at the extension where your modem is connected. Your modem will work the same way it always has. You don't have to change your dial-up settings for the modem.

2. If you use a telephone and modem on the same extension, do not enable direct line access**.** A TalkSwitch extension with direct line access immediately connects to one of your telephone lines when you pick up the telephone's handset, preventing you from accessing other system features without first pressing **Flash** or **Recall** on an analog extension or ****** on a IP extension.

   You can have your modem access a specific telephone line using a hunt group.

   Change your modem's dial properties to include the intended hunt group — **9** (**0** in some regions) or **81**–**88**, followed by the letter **w**, before the dial-up networking phone number (e.g. **9w789-1234**). The letter **w** after your dial-out access number tells your modem to dial the access number 9, wait until it detects a dial tone, and then dial your dial-up networking number.

   If your modem has problems dialing out, use a comma (,) to create a 2 second pause instead of the **w** (wait for dial tone).

Because TalkSwitch units are digital systems, modem calls will only operate at a maximum speed of approximately 28 Kbps.

FORWARDING CALLS OUT OF THE OFFICE

Automatic call forwarding

The auto attendant can forward an office call to a remote extension. Use the auto attendant message to prompt callers to select your remote extension number(s). (e.g. *"To speak to John Smith, dial 211."*)

Manual call forwarding

If you have answered a call at a local extension, you can forward the call to a remote extension.

1. Put the call on hold at the local extension.

2. Dial the remote extension number (e.g. 211) to which you want the call forwarded.

3. Hang up an analog extension or press **Transfer** or **xfer** on an IP extension.

The call is handled according to the call cascade options of that remote extension.

Conditional call forwarding

If the auto attendant answers a call and the caller selects an extension or ring group number that is busy or is not answered, the auto attendant can forward the call to a remote extension. Conditional call forwarding is configured in the TalkSwitch software using the call cascade's *No answer* or *Busy* settings for each local extension, remote extension and ring group.

Transferring calls from a remote extension

You can forward calls from a remote extension without using Centrex or 3-way calling services. Calls can be transferred to any local or remote extension, ring group or voice mailbox. This feature also applies to VoIP numbers.

1. Press ****** to place a call on hold at a remote extension anytime during a conversation. You hear the same dial tone as at a local extension and have the following three possibilities:
    - Press ****** to retrieve the call placed on hold.
    - Dial any local extension, remote extension, voice mailbox, VoIP location or ring group.
        - If the system is configured to perform a blind transfer, the system plays *"Call Transferred — Goodbye."* and then hangs up.
        - If the system is configured to allow call screening, determine whether the other person wants the call, and then press ****4** to complete the transfer or ****5** to cancel the transfer and return to the caller.

Screening options for forwarded calls

The screening options for call forwarding are configured in the TalkSwitch software using the *Answered* tab in the **Remote Extensions** page.

Forwarding calls with screening

When *play accept/reject prompt* is configured in the *Answered* call cascade option in the **Remote Extensions** page, TalkSwitch plays a pre-recorded prompt to callers asking them to hold and then dials your call forwarding number. When you answer the forwarded call, you hear a prompt saying: *"You have a forwarded call. To accept the call, press* **#**. *To reject the call, press* *****." If you reject the call, it is routed according to the call cascade option specified next to *If a call is rejected from this ext.*

When *play caller's name first* is configured in the *Answered* call cascade option, TalkSwitch plays a pre-recorded prompt to callers asking them to record their name at the sound of the tone. TalkSwitch asks the caller to hold and dials the call forwarding phone number. When you answer the forwarded call, TalkSwitch plays a pre-recorded prompt saying: *"This is call forward, you have a call from..."* and plays the recording of the caller's name. To accept the forwarded call, press **#** on your telephone keypad, re-route the call to its call screening call cascade options by pressing *****.

When either screening option is used, music on hold is enabled and you are not using the 3-way calling feature (*Same Line Connect*) to forward calls, the caller hears music while TalkSwitch is waiting for the remote extension to accept the call.

If a forwarded call is not answered or the line is busy, TalkSwitch performs the following, based on how the call was originally forwarded:

- If the call was automatically forwarded, it follows the remote extension's *No Answer* or *Busy* call cascade action.
- If the call was manually forwarded, it follows the remote extension's *No Answer* or *Busy* call cascade action. This action can be changed to ring the extension back to the person who performed the transfer.
- If the call was conditionally forwarded, it follows the call cascade action in the *No Answer* or *Busy* for the original extension selected.

CALLS OVER VOIP WITH IP EXTENSIONS AND GATEWAYS

If teleworkers are connected to the system via a certified compatible IP device (i.e. gateway or IP extension), there are several ways for them to interact with TalkSwitch.

Making calls

1. Ensure the VoIP number is programmed with an auto attendant.
2. Dial the VoIP number.
3. When the auto attendant answers, dial an extension, voice mailbox, VoIP number or hunt group.

Transferring calls

1. If the call is routed via TalkSwitch, you can transfer the caller by dialing ** and any extension or voice mailbox.
2. If the call came directly from a IP gateway or phone, you can transfer the call if your phone or service provider supports this feature.

USING THE VOICEMAIL SYSTEM

The recording time for internal music on hold, voicemail, and the auto attendants is shared on the TalkSwitch unit. You can expand the built-in memory using memory cards.

Note that the system allows up to 99 voicemail messages per mailbox. Once a mailbox fills up, callers won't be able to leave voicemail messages for that user. Therefore users should delete voicemail messages before the mailbox fills up.

Activating voice mailboxes

You can activate voicemail with a mailbox, as an announcement, or you can deactivate it. There are three types of voice mailboxes:

- Local extension mailboxes are associated with local extensions. All local extensions are activated with a respective mailbox by default.
- Remote extension mailboxes are associated with the remote extensions. If you activate a remote extension, the mailbox is activated by default.
- General voice mailboxes are not associated with extensions.

Callers can reach voice mailboxes from a call cascade, be transferred from a local extension or have announcements played through the auto attendant.

If you want incoming calls to go straight to a voice mailbox after a specified number of rings, program an auto attendant to answer the line and then route the call to voicemail.

To leave a message from an analog extension:

1. Press * and the mailbox number.

To leave a message from an IP extension:

1. Press * and the mailbox number and press **Dial, Send** or **#**.

To transfer a caller directly to voicemail from an analog extension:

1. Press **Flash** or **Recall** *, the mailbox number and hang up.

To transfer a call to a voice mailbox from an IP extension:

1. Press **Transfer** or **xfer** *.
2. Enter the mailbox number.
3. Press **Transfer** or **xfer** to complete.
4. Press **Dial, Send** or **#**.

When a caller presses **#** after leaving a message or after 2 minutes have elapsed, TalkSwitch plays a prompt asking callers if they wish to keep the message, listen to the message or re-record the message. Callers can dial **0** (**9** in some regions) and the system directs the call according to the settings of that mailbox.

Retrieving messages and accessing a voice mailbox

The first time you access your voice mailbox, you are prompted to assign a password, record a greeting and record your name for the dial-by-name directory.

For the general mailboxes, there are no directory listings and you are not prompted to record a name. If you wish to have your name in the company directory, use the mailbox associated with its extension, as the directory is related to the extension's mailbox. When your mailbox has been set up, you access it through the prompt instructions.

If a new message is left in a local extension mailbox, TalkSwitch plays a stutter dial tone when the handset is picked up (not applicable to the TS-850i). If the extension supports a frequency-shift keying (FSK) message waiting indicator (not applicable to IP extensions), the light flashes. The TS-80, TS-200, TS-400 and TS-600 extensions support a message waiting counter. The display on the extension shows the number of new messages stored in the mailbox(es) associated with that extension.

1. If direct line access is enabled, be sure to press **Flash** or **Recall** before trying to access your mailbox. From a Grandstream or Polycom IP extension, press **MSG** or **Messages** button.
2. Press ****#** from a local extension or ****** and the mailbox number at the auto attendant if you are dialing from an outside location. From an IP extension, press **Dial, Send** or **#**.
3. Dial **1** to listen to messages.

   By default, TalkSwitch starts with newest message before it plays the older ones. While the message is playing or after the message has finished, you can dial the following:

1	to rewind 10 seconds
11	to rewind to beginning of the message
3	to skip ahead 10 seconds
33	to skip to the end of the message
5	to listen to the time and date stamp
6	to forward the message
7	to delete a message
9	to save a message
*	to back up one level
#	to skip to next message, leaving new messages as new.

4. Dial **2** to change greeting options.

   This allows you to record a new personal voicemail greeting. The default greeting is: *"The extension you have reached is unavailable at this time. Please leave a message after the tone".*

   While recording your greeting or announcement, remember that callers can press ***** to return to the auto attendant, if the call came from the auto attendant. When they are done recording, they can press **#** for more options. You may want to give your callers these options for their convenience. In local extension mailbox options, **0** (**9** in some regions) can be configured as an option for additional routing.

5. Dial **3** to change personal options.

   - change your current password
   - turn the auto date and time stamp on/off
   - set up or change pager and remote phone notification

   By default, the auto date and time stamp is on. This information is played at the end of each message. If you have caller ID service, the phone number is displayed with the time and date.

6. Dial **4** to record your name for the dial-by-name directory.

   If you haven't recorded your name for the directory or if you wish to change the recording, dial **4**. The dial-by-name directory is accessed at the auto attendant according to your configuration. Callers are directed to enter the first 3 letters of the employee's last name. The system finds the exact or closest match and plays the recorded name with the extension number, giving callers the option to connect to that extension.

You must record your name in order for the extension to be included in the dial-by-name directory.

7. You can transfer a caller to voicemail by pressing **Flash** or **Recall **** and entering the mailbox number. From an IP extension, press **Transfer** or **xfer** + mailbox + **Dial, Send** or **#**.

Recording an announcement on a local extension

An announcement is a recorded message providing information to callers. It can be recorded using your phone as a voicemail message or loaded as an audio file. The system hangs up after playing an announcement. Callers have the option of pressing ***** during an announcement to return to the auto attendant. Callers can also dial **0** (**9** in some regions) to perform other actions if this is configured in the voicemail options.

You might include in the announcement a mention of the selections a caller may press and leave a section of silence at the end to give the caller time to make a selection.

To record an announcement, access a voice mailbox and select option **2** to change greeting options as described in *Retrieving messages and accessing a voice mailbox* on page 220.

Pager and cell phone notification

When a new message is left in a voice mailbox, it can notify a pager, cell phone or a remote number.

You can configure the system to notify you for every new message or for only the first new message received since the last time you accessed new messages.

If you use remote phone notification, you can accept or reject the notification call when you answer.

- Press **#** to listen to messages.
- Press ***** to postpone listening to messages.

MUSIC ON HOLD

TalkSwitch can play music to callers when they are on hold, parked or queued. The music played is provided by the source you have connected to the music jack or from a sound file (8 KHz, 8 bit, mono, u-law, .wav format) stored on the unit(s). When a sound file is loaded on one of the TalkSwitch units, it is duplicated to all units on the LAN.

Music on hold and call forwarding to remote extensions

If you are using the telephone company's 3-Way Calling/Conference service with the same line connect feature to activate the TalkSwitch call forwarding feature, callers do not hear music on hold while they are being forwarded to a remote extension. The caller is put on hold by the telephone company's central office switch rather than by TalkSwitch, and the caller hears silence while being transferred.

A system prompt indicates that callers will hear silence while the system is locating their party. This prompt explains to callers why there is silence for an extended period of time before connecting to the remote extension. This prompt cannot be disabled.

If you have two or more units on a LAN, we recommend that you use a .wav file. If you use an external audio source, connect it to each TalkSwitch unit. Use 1/8" audio splitters, which are available at most electronics stores. See *Unit back panel* on page 4 for more details.

SWITCHING MODES

Ensure the TalkSwitch management software is closed during automatic mode switching, or if an administrator is switching modes remotely. In these cases, the mode cannot switch while the TalkSwitch management software is open.

Note that the mode can switch while the TalkSwitch management software is open, if you are switching modes using the software.

Switching modes manually

Use telephone keypad commands at a local extension or at the auto attendant to switch modes.

1. Enter command mode by pressing **#** on an analog extension phone, or ***55#** on a TalkSwitch IP phone (note: other brands may use ***55 Send** or ***55 Dial**).
2. Enter the system password, followed by **#**.
3. Enter one of the following commands:
   - **30#** to check the current mode.
   - **31#** to activate Mode 1.
   - **32#** to activate Mode 2.
   - **33#** to activate Holiday Mode.

Switching modes automatically

Use the TalkSwitch management software to set up automatic mode switching. See *Scheduling* on page 31.

If you switch modes manually, TalkSwitch will automatically switch modes during the next scheduled mode change.

USING CALL BRIDGE AND CALL BACK

Working together, call back and call bridge act as your personal long-distance operator. Whether you are across town or around the world, these two features allow you to place calls from your office telephone line.

You can make a call to TalkSwitch, access a telephone line connected to TalkSwitch and enter the number you want to dial. This is especially useful when you are out of the office with your cell phone and need to make a long-distance call. You can avoid the long-distance cell rates by making a local call to TalkSwitch and access your office savings plan through call bridge.

Using call bridge

1. Dial into a telephone line.
2. When the auto attendant answers, select a hunt group — **9** (**0** in some regions), **81**–**88** or *Same Line Connect* **80**.
3. Enter your account password. If you are using *Same Line Connect* and your line supports the 3-way calling/conference service, you are prompted to enter the phone number.
4. When you have completed your call, do one of the following:

   a) Make another call by pressing **##**.

   TalkSwitch disconnects you from the call in progress. You can dial another number or redial the same number.

   b) Activate the auto attendant by pressing **#***.

   TalkSwitch ends the call bridge session and activates the auto attendant. When the auto attendant message begins to play, you have the following options:

   - Press **#** to enter command mode
   - Dial **6** to change/enter call back settings
   - Dial a local or remote extension or a ring group. This option allows you to contact someone in your office, check your voicemail, etc.

   c) End the call bridge session by hanging up. This disconnects you from TalkSwitch. To ensure TalkSwitch disconnects at the end of a call bridge session, press **##** before hanging up.

The call bridge phone number you dial can be local, long-distance, toll free (800 and 888) or international. Enter the phone number as you would in your office. For long-distance calls, include the 1, country code and area code.

Using call back

The call back feature allows you to initiate TalkSwitch to call you at a specified phone number. This gives you access to the following:

- Call bridge
- Local extensions, remote extensions and ring groups
- Voicemail
- Configuration settings

Using call back involves 3 steps:

1. Call your TalkSwitch phone number to activate call back.
2. Answer and accept call back. TalkSwitch calls you back within 30 seconds.
3. At the auto attendant you can choose to perform call bridge, dial an extension, access and retrieve voicemail or configure TalkSwitch.

Activating call back

There are two ways to activate call back from your out-of-office location.

Using prompted call back activation

1. Dial a TalkSwitch phone number.
2. Let the line ring until the auto attendant answers.
3. To initiate call back at the current call back number, dial **61** on the telephone keypad, and then hang up. The system will call you back and then prompt you for your password.
4. To initiate call back at a new call back number, dial **62** on the telephone keypad. The system will prompt you for your password. Enter your password and the new call back number, and then hang up. The system will call you back and then prompt you for your password.

For prompted call back, TalkSwitch always dials the last prompted call back number entered. If you do not have the auto attendant answer a line, a generic TalkSwitch auto attendant will answer after 15 rings. You can then dial **6** to access the call back settings.

Using auto call back activation

1. Dial one of the call back phone numbers that has been configured for auto call back activation.
2. Let the line ring at least once but no more than three times and hang up before the call is answered.
3. Within 20 seconds, TalkSwitch calls your auto call back number.

It is important to remember that all call back settings must be configured to use auto call back. Please ensure the auto attendant for the line using call back does not answer before 4 rings.

Accepting the call back

There are two ways to accept a call back from TalkSwitch.

Directly answering the call back

1. Answer the call and press **#**.

   If required, enter your password. This option is used when you have a direct line to your call back phone number.

Using the announced message option

The *Use announced message* option allows the call back from TalkSwitch to reach you when you are in a hotel or an office where calls are intercepted by a receptionist or switchboard operator. When the call back is answered, TalkSwitch begins playing your pre-recorded message (e.g. *"Please forward this call to Bob Smith in room 312"*). The message is repeated for 2 minutes.

1. When the call is forwarded to you, press **#** and enter your password on the telephone keypad to accept the call back. If the call back is not accepted within the initial 2-minute period, TalkSwitch disconnects the call back.

When you accept the call, you hear the auto attendant. You can select to check your voicemail, ring an extension, use call bridge or enter command mode.

CHAPTER 4

CALL DETAIL RECORD LOGGING

INTRODUCTION

This chapter covers the setup and retrieval of call detail records (CDRs), which are records generated by calls through the phone system.

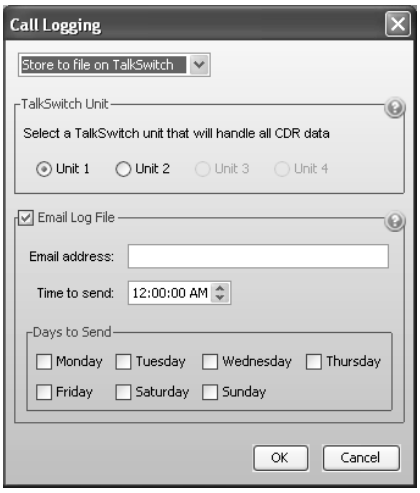
The TalkSwitch unit can store the CDRs. The system can automatically e-mail the CDRs, or you can manually retrieve them with the Call Reporting licensed add-on application or a web browser.

If multiple units are networked on a LAN, the CDRs from all the units are stored on one unit. The TalkSwitch unit stores approximately 6000 records in a circular buffer. This means that after about 6000 calls, the oldest records will be deleted and new ones created. On a very busy system, 6000 records may only cover a couple of days, but on a system with less traffic, 6000 records may cover more than a month.

The TalkSwitch unit can instead send the CDRs to a computer in real-time over the USB port. In this case the TalkSwitch unit does not store any CDRs.

ENABLING CALL DETAIL RECORD LOGGING

1. Choose **Tools > Call Logging Output (CDR)**. The **Call Logging** window appears.



2. Select how the TalkSwitch unit will process CDRs. Choices are:

   - *Deactivated* — CDRs are neither stored nor sent to the computer.
   - *Real-time to Serial/USB port* — CDRs are sent to the computer in real-time. Selecting this enables the *TalkSwitch Unit* area.
   - *Store to File on TalkSwitch* — CDRs are stored in the TalkSwitch unit. Selecting this enables the *TalkSwitch Unit* area and the *Email Log File* area.

TalkSwitch unit

The *TalkSwitch Unit* area allows you to select the TalkSwitch unit that will process the CDRs. It is enabled if you selected *Real-time to Serial/USB port* or *Store to File on TalkSwitch*.

1. Select the TalkSwitch unit that will process the CDRs. Only active units are available.

Email log file

The *Email Log File* area allows you to have the system regularly e-mail a log file containing the CDRs. It is enabled if you selected *Store to File on TalkSwitch*.

Note that if you want the system to e-mail a log file, you must configure the outgoing mail server (SMTP). See *Email Service* on page 40.

1. To have the system e-mail the log file, select the *Enabled* check box.
2. Enter the *Email Address*.
3. Select the *Time to send*.
4. Select the *Days to send*.

RETRIEVING CALL DATA RECORDS

There are two methods for retrieving CDRs:

- Call Reporting licensed add-on application
- Browser

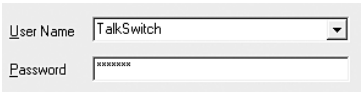
Call reporting

The Call Reporting licensed add-on application automatically connects to the TalkSwitch unit, downloads the CDRs, and builds detailed reports of call activity. The application requires the *Store to File on TalkSwitch* setting in the **Call Logging** window.

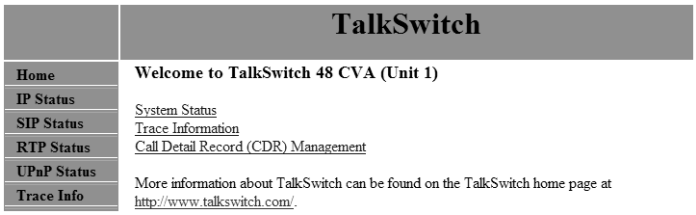
Browser

You can connect to the TalkSwitch unit and download the CDRs using a browser. Retrieving the CDRs using a browser requires the *Store to File on TalkSwitch* setting in the **Call Logging** window.

1. Select the **IP Configuration** page in the TalkSwitch management software, and note the *Unit IP Address* (e.g. 192.168.1.102).
2. Select the **Administration** page in the TalkSwitch management software, and note the *System name*. You will also require the *System password*.
3. Enter the following into the *Address* field of a web browser: **http://**, followed by the unit IP address, followed by **:8484/** (e.g. **http://192.168.1.102:8484/**). The **Log in** window appears.

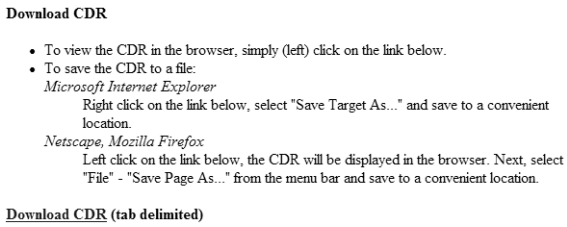


4. Enter the system name in the *User Name* field of the **Log in** window, and the system password in the *Password* field. If the system name is blank, enter **talkswitch**. The browser shows the **Welcome to TalkSwitch** screen.



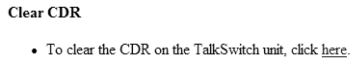
5. Click the **Call Detail Record (CDR) Management** link. The browser shows the **Call Detail Record (CDR) Management** screen.

6. Scroll down to the **Download CDR** section.


Download CDR
To view the CDR in the browser, simply (left) click on the link below.
To save the CDR to a file:
Microsoft Internet Explorer
Right click on the link below, select "Save Target As..." and save to a convenient location.
Netscape, Mozilla Firefox
Left click on the link below, the CDR will be displayed in the browser. Next, select "File" - "Save Page As..." from the menu bar and save to a convenient location.
Download CDR (tab delimited)

7. You can either right-click the **Download CDR** link to save the information to your PC, or can click the link to view the current information.

8. Once the CDR file has been downloaded, scroll down to the **Clear CDR** section.



ANALYZING THE DATA

We recommend that you clear the TalkSwitch unit immediately after downloading the CDRs. The unit will then start saving data to a new file. This allows easier viewing and analyzing of the captured information.

The data can be viewed either through the web interface or downloaded into a spreadsheet (e.g. Excel) as a tab delimited text file. The log is divided into 13 columns identified by numbers:

1. Type	5. Time	9. Caller ID Name
2. Log	6. Duration	10. Line
3. Event	7. Connection	11. Account Number
4. Date	8. Phone Number	12. User

Each column contains specific information related to the current state of the call. The following is a list of the columns with a description:

1. **Type** — This column can contain the following characters:

   - **I** — Inbound call

- **T** — Transfer state
- **O** — Outbound call
- **B** — Bridged call. For example, a call in on one line that is forwarded over another line. This can include calls forwarded to remote extensions.
- **X** — Blocked call
- **A** — Account number assigned to the call with the same log #.

2. **Log** — This column indicates log number for the call. For the duration of the call, the log number remains the same. The event number will increment with each change of state. The first digit of the log number indicates on which unit the event originated.
3. **Event** — This column tracks various state changes of the call. For example, when an auto attendant answers, the counter might be 1. When the caller selects an extension, the counter increments to 2, etc.
4. **Date** — This column shows the date of the call (MM/DD/YYYY).
5. **Time** — This column contains the time that the call entered a specific state. The time is displayed in 24-hour format (HH:MM).
6. **Duration** — This column indicates the total time the call was in this state/event. The time is displayed in 24-hour format (HH:MM:SS).
7. **Connection** — This column indicates where the call was for each event. This includes auto attendants, voicemail, extension ringing or queuing.
   - Exxx = Connected to extension xxx (xxx = 111–148, 151–188)
   - A00x = Connected to auto attendant x (x= 1 to 9)
   - Rxxx = Ringing at extension xxx (xxx = 111–148, 151–188)
   - RC0 = Ringing at telephone line.
   - Qxxx = Queued at extension xxx (xxx = 111–148, 151–188, 300–309)
   - Hxxx = Call on hold at extension xxx (xxx = 111–148, 151–188)
   - Mxxx = Connected to voice mailbox xxx (xxx = 111–148, 151–188, 211–248, 410–449)
   - Cxyy = Forwarded out on unit x line yy. (See "10. Line" below for more details).
8. **Phone Number** — This column displays the phone number of the inbound or the outbound caller if available.
9. **Caller ID Name** — This column displays the name of the caller if available.
10. **Line** — This column indicates on which unit and line the call came in or out. The format of this information is unit and line number. Lines 01–08 are telephone lines and 09–16 are VoIP lines.

    Example: 101 — unit 1, line 1
    203 — unit 2, line 3
    110 — unit 1, VoIP 2

11. **Account Number** — This column contains an account number that the user can assign after a call. This allows you to group calls together that may have been placed over time to

calculate the total amount of time spent with a particular customer. The account number can be a numerical code up to 15 digits long.

To assign an account number after a call has been completed, dial ***88**. When prompted, dial <Account Number> + **#**.

If **Direct Line Access** is enabled on an extension:

- Analog extension — press **Flash**, then dial ***88**. When prompted, dial <Account Number> + **#**.
- IP extension — press ******, then dial ***88**. When prompted, dial <Account Number> + **#**.

12. **User** — If Permissions are enabled, this column shows the code name associated with the access code used to dial out.

The data can be imported into any call management software or an Excel spreadsheet as a tab delimited text file. After the import, the data can be grouped and tallied based on the criteria set forth.

Example:

1. Total line usage for a day	4. Average call times
2. Number of inbound calls	5. Number of calls per line
3. Wait times	

CHAPTER 5

VOIP INFORMATION

ABOUT VOIP

VoIP-enabled TalkSwitch units route VoIP calls over the Internet or private IP network, and other calls over the traditional telephone network.

VoIP-enabled TalkSwitch units uses the industry-standard Session Initiation Protocol (SIP) for VoIP calls. In this guide the term SIP is often used in place of VoIP.

There are three ways to use VoIP. A system can use one, two or all three methods.

- External IP extensions
- TalkSwitch VoIP network
- Service provider VoIP network

About SIP servers

A VoIP network requires a SIP server to manage calls across the network. The SIP server contains registration information for all SIP devices in the network. When making a VoIP call, a device will contact the SIP server to get contact information for the destination. The device will then establish a connection with the destination. The SIP server is also involved with call termination.

All VoIP-enabled TalkSwitch units have built-in SIP servers to manage a TalkSwitch VoIP network, but you must designate only *one* TalkSwitch unit as the SIP server. TalkSwitch systems at other locations are configured as SIP clients.

Note: For VoIP calls made through a VoIP service provider, the service provider is the SIP server.

Selecting a location for the SIP server

Any VoIP-enabled TalkSwitch unit on the network can be designated as the SIP server.

To simplify network operation and reliability, we recommend you choose a location that has a static, rather than a dynamic, public IP address. A static public IP address does not change, providing a consistent connection address for each location to communicate with the SIP server. See *Setting the public IP address* on page 234.

CONNECTING TO A NETWORK

Each location requires a high-speed connection to the Internet or private IP network that is sufficient for VoIP calls. This is required for a location using an external IP extension, TalkSwitch VoIP network, or service provider VoIP network.

Connecting to local and IP networks

Set up a LAN (local area network) with an Ethernet switch, router, and modem or gateway. The Ethernet switch can be integrated into the router. Connect a computer to the LAN for configuring the TalkSwitch unit and other equipment. Optional IP phones can be connected as well. Use the provided RJ-45 Ethernet cable to connect the TalkSwitch unit to the LAN. To ensure reliability, all equipment should be connected to a UPS (uninterrupted power supply).

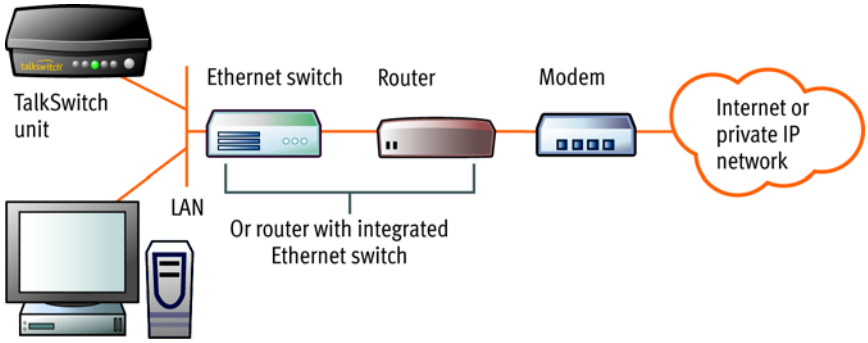


Figure 17: Local and IP network setup

Connecting a VoIP gateway

In order to connect to a TalkSwitch VoIP network, a location can use a SIP-compatible VoIP gateway instead of a VoIP-enabled TalkSwitch unit. For example, the Mediatrix® 2102 is a TalkSwitch-certified third-party VoIP gateway that is suitable for small branch offices or teleworkers.

Confirming network capacity

Ensure the LAN in each location has a reliable high-speed broadband connection to the Internet or private IP network. The quality of VoIP calls, especially on mixed voice and data networks, depends on high data-transfer rates across the network. The limiting factor is the upstream bandwidth to the ISP (Internet Service Provider). 'Lite' broadband connections (128 Kbps or less) are not suitable for simultaneous voice and data traffic.

To test the VoIP call capacity based on your Internet connection speed, perform the test found in the support section at **http://global.talkswitch.com**.

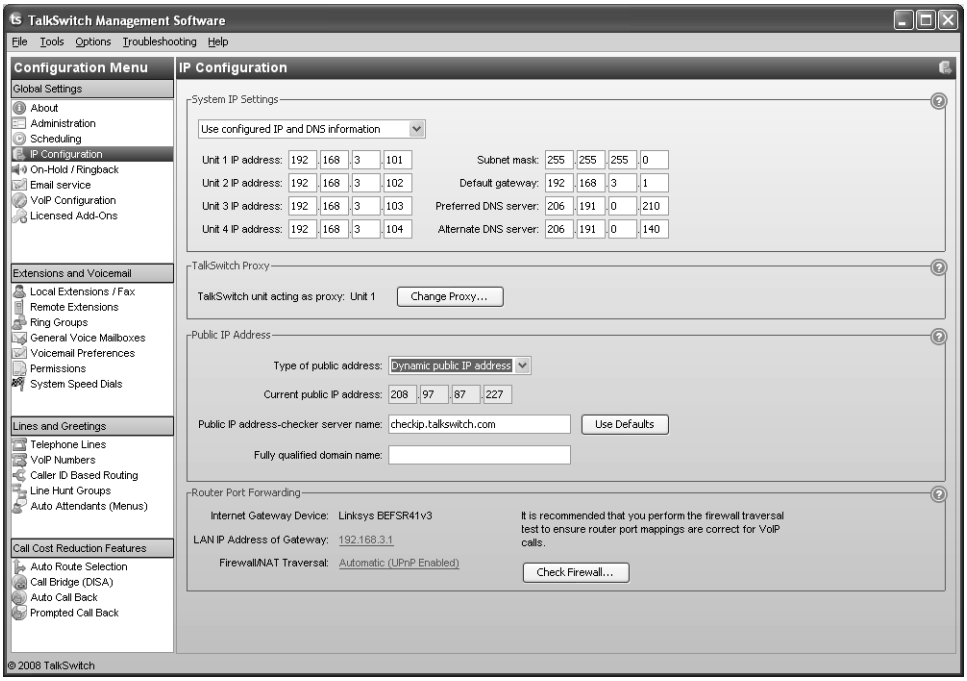
CONFIGURING IP ADDRESSES

Starting the TalkSwitch management software

1. Double-click the **TalkSwitch Configuration 6.12** icon on your desktop. You can also select **Start > Programs > TalkSwitch 6.12 > TalkSwitch Configuration 6.12**. The **TalkSwitch Management Software** window appears, and the software attempts to detect your TalkSwitch unit.
2. Select your language. You can select *English*, *Français* or *Español*.
3. Once the software detects your TalkSwitch unit, click **Configure Auto-Detected TalkSwitch System**. The **About** page appears.

Setting the system IP settings

Each TalkSwitch unit must be configured with a local IP address. These addresses are used to direct VoIP calls to the appropriate location.


TalkSwitch Management Software
File Tools Options Troubleshooting Help
Configuration Menu
Global Settings
About
Administration
Scheduling
IP Configuration
On-Hold / Ringback
Email service
VoIP Configuration
Licensed Add-Ons
Extensions and Voicemail
Local Extensions / Fax
Remote Extensions
Ring Groups
General Voice Mailboxes
Voicemail Preferences
Permissions
System Speed Dials
Lines and Greetings
Telephone Lines
VoIP Numbers
Caller ID Based Routing
Line Hunt Groups
Auto Attendants (Menus)
Call Cost Reduction Features
Auto Route Selection
Call Bridge (DISA)
Auto Call Back
Prompted Call Back
IP Configuration
System IP Settings
Use configured IP and DNS information
Unit 1 IP address: 192 168 3 101
Unit 2 IP address: 192 168 3 102
Unit 3 IP address: 192 168 3 103
Unit 4 IP address: 192 168 3 104
Subnet mask: 255 255 255 0
Default gateway: 192 168 3 1
Preferred DNS server: 206 191 0 210
Alternate DNS server: 206 191 0 140
TalkSwitch Proxy
TalkSwitch unit acting as proxy: Unit 1
Change Proxy...
Public IP Address
Type of public address: Dynamic public IP address
Current public IP address: 208 97 87 227
Public IP address-checker server name: checkip.talkswitch.com
Use Defaults
Fully qualified domain name:
Router Port Forwarding
Internet Gateway Device: Linksys BEFSR41v3
LAN IP Address of Gateway: 192.168.3.1
Firewall/NAT Traversal: Automatic (UPnP Enabled)
It is recommended that you perform the firewall traversal test to ensure router port mappings are correct for VoIP calls.
Check Firewall...
© 2008 TalkSwitch

1. Select the **IP Configuration** page. By default, *Obtain IP and DNS information automatically* is selected and the area shows IP addresses from the router.

2. Change *Obtain IP and DNS information automatically* to *Use configured IP and DNS information* in order to lock in the IP addresses.

3. In some cases, the *System IP Settings* area may be blank because your router has not delivered the IP addresses. If so, enter the following IP addresses from your LAN administrator:

   a) Enter a static IP address for each TalkSwitch unit in the *Unit IP address* boxes.

   b) Enter the *Subnet mask* for the LAN. This address determines the subnet that the unit IP addresses belong to.

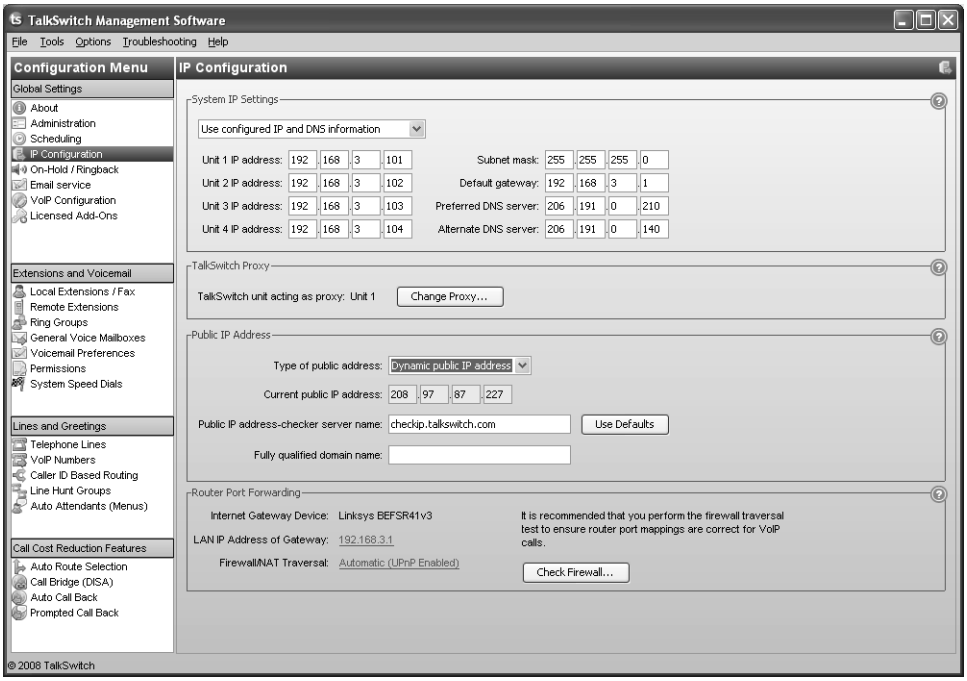
   c) Enter the IP address of the *Default gateway* on your network. A gateway is a hardware device that connects the office network to the Internet. The router may act as default gateway.

   d) Enter the IP address of the *Preferred DNS server*. DNS is a service that is used to resolve a domain name to an IP address. The router may act as DNS server.

   e) If applicable, enter the IP address of the *Alternate DNS server*.

Setting the public IP address

If you are setting up an external IP extension, or the SIP server of a TalkSwitch VoIP network, you must configure a public IP address for the TalkSwitch system. Some service provider VoIP networks also require the TalkSwitch system to have a public IP address.


TalkSwitch Management Software
File Tools Options Troubleshooting Help
Configuration Menu
Global Settings
About
Administration
Scheduling
IP Configuration
On-Hold / Ringback
Email service
VoIP Configuration
Licensed Add-Ons
Extensions and Voicemail
Local Extensions / Fax
Remote Extensions
Ring Groups
General Voice Mailboxes
Voicemail Preferences
Permissions
System Speed Dials
Lines and Greetings
Telephone Lines
VoIP Numbers
Caller ID Based Routing
Line Hunt Groups
Auto Attendants (Menus)
Call Cost Reduction Features
Auto Route Selection
Call Bridge (DISA)
Auto Call Back
Prompted Call Back
© 2008 TalkSwitch
IP Configuration
System IP Settings
Use configured IP and DNS information
Unit 1 IP address: 192 168 3 101
Unit 2 IP address: 192 168 3 102
Unit 3 IP address: 192 168 3 103
Unit 4 IP address: 192 168 3 104
Subnet mask: 255 255 255 0
Default gateway: 192 168 3 1
Preferred DNS server: 206 191 0 210
Alternate DNS server: 206 191 0 140
TalkSwitch Proxy
TalkSwitch unit acting as proxy: Unit 1
Change Proxy...
Public IP Address
Type of public address: Dynamic public IP address
Current public IP address: 208 97 87 227
Public IP address-checker server name: checkip.talkswitch.com
Use Defaults
Fully qualified domain name:
Router Port Forwarding
Internet Gateway Device: Linksys BEFSR41v3
LAN IP Address of Gateway: 192.168.3.1
Firewall/NAT Traversal: Automatic (UPnP Enabled)
It is recommended that you perform the firewall traversal test to ensure router port mappings are correct for VoIP calls.
Check Firewall...

1. Set the *Type of public address*. Choices are:

   - *Dynamic public IP address* — This is the default setting. Your ISP (Internet Service Provider) will assign different public IP addresses to your location. The TalkSwitch system will check its public IP address every few minutes. When the public IP address changes, the TalkSwitch system will automatically use the new one, in order to manage VoIP calls properly.

     If *Dynamic public IP address* is selected, the window completes the *Current public IP address*, and sets *Public IP address-checker server name* to **checkip.talkswitch.com**. This server runs an IP checker application that provides public IP addresses to TalkSwitch systems.

   - *Static public IP address* — A static IP address is fixed. Your Internet Service Provider assigns the static IP address. If this location will have the SIP server for a TalkSwitch VoIP network, we recommend using a static IP address.

     If *Static public IP address* is selected, the window allows you to enter the *Current public IP address*, and disables the *Public IP address-checker server name* box.

2. If you selected *Dynamic public IP address*, enter the *Fully qualified domain name* (FQDN). Get the FQDN from your ISP.

   A DDNS (Dynamic Domain Name Service) provider such as **www.dyndns.com** matches your dynamic IP addresses to your FQDN, so your TalkSwitch VoIP network or external IP phones will continue to work when the IP address changes.

If your router supports DDNS, ensure it supports your DDNS provider, and configure it to update the DNS servers.

If your router does not support DDNS, download one of the applications specified on **www.dyndns.com**. To update the DNS servers, the application needs to run on a PC connected to the same LAN as the TalkSwitch system.

3. If you selected *Static public IP address*, enter the *Current public IP address* from your Internet Service Provider. Leave the *Fully qualified domain name* box blank.

   If the TalkSwitch unit is not behind a router, or if a private virtual network is used, the public IP address is the local IP address of the TalkSwitch unit acting as local proxy.

   Note that it will take up to one minute for the new static public IP address to take effect.

4. Record the *Current public IP address* or the *Fully qualified domain name*.

   If you are setting up an external IP extension, you will need the public IP address or FQDN of the TalkSwitch system.

   If you are setting up a TalkSwitch VoIP network, the TalkSwitch profiles of the SIP clients must be configured with the public IP address or FQDN of the SIP server.

CONFIGURING THE ROUTER

The *Router Port Forwarding* area of the **IP Configuration** page displays the type of gateway device (i.e. the type of router), the IP address of the gateway (i.e. router), and whether router configuration is required.

A router is a gateway between the local area network and the Internet. Most routers have a firewall to block unwanted data from the Internet. For voice data to reach the TalkSwitch system through the firewall, port forwarding is required. Port forwarding allows the router to map ports to the IP addresses of the TalkSwitch units. Valid Internet data will use the ports to go through the firewall to the TalkSwitch units.

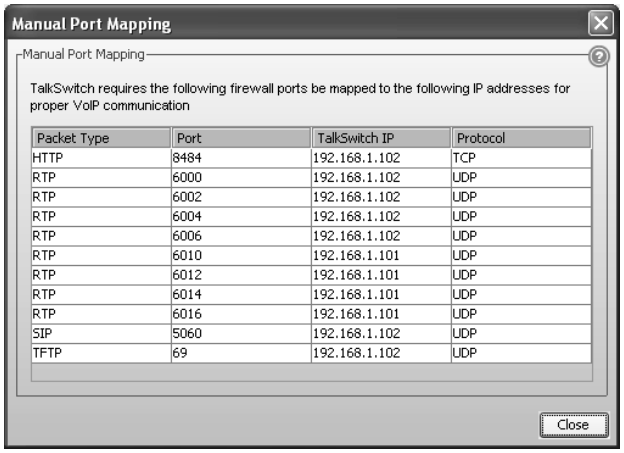
If you are setting up external IP extensions, a TalkSwitch VoIP network, or a VoIP service that doesn't handle port forwarding, port forwarding is required.

If port forwarding is required, and your router supports uPNP (Universal Plug and Play), ensure uPNP is enabled. The TalkSwitch system will use uPNP to automatically set up port forwarding, and the **Automatic (uPNP Enabled)** link will appear. No router configuration is required.

If port forwarding is required but your router doesn't support uPNP, or automatic port forwarding doesn't work, the **Manual Port Mapping Required** link will appear. You will need to configure the router as described below.

1. Select the **IP Configuration** page.

2. If required, click the **Manual Port Mapping Required** link. The **Manual Port Mapping** window appears. It lists the packet type, port number, IP address and protocol of each required port.


Manual Port Mapping
Manual Port Mapping
TalkSwitch requires the following firewall ports be mapped to the following IP addresses for proper VoIP communication
Packet Type | Port | TalkSwitch IP | Protocol
HTTP | 8484 | 192.168.1.102 | TCP
RTP | 6000 | 192.168.1.102 | UDP
RTP | 6002 | 192.168.1.102 | UDP
RTP | 6004 | 192.168.1.102 | UDP
RTP | 6006 | 192.168.1.102 | UDP
RTP | 6010 | 192.168.1.101 | UDP
RTP | 6012 | 192.168.1.101 | UDP
RTP | 6014 | 192.168.1.101 | UDP
RTP | 6016 | 192.168.1.101 | UDP
SIP | 5060 | 192.168.1.102 | UDP
TFTP | 69 | 192.168.1.102 | UDP
Close

3. To access the router configuration:

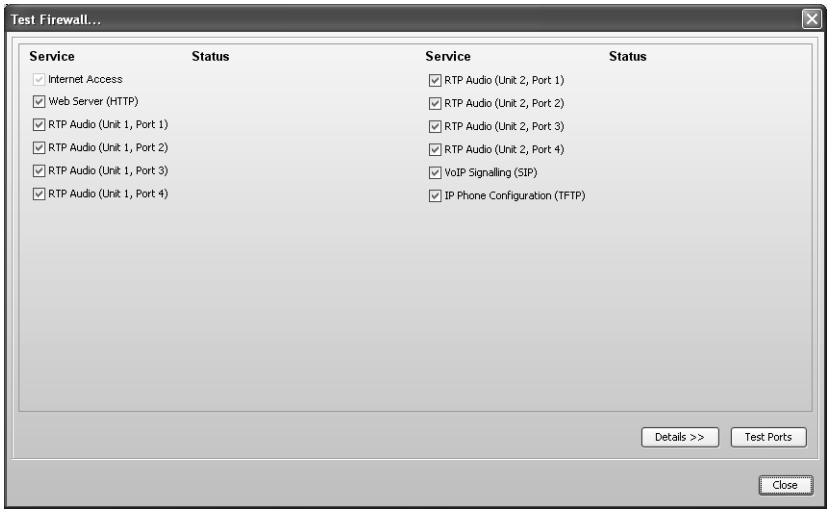
   a) Click the link containing the IP address of the gateway. The default browser starts, and prompts you for the router's user name and password.

   b) Enter the router's user name and password. The browser shows a setup screen.

   c) Navigate to the screen used to set up port forwarding. See your router documentation.

   d) Set up port forwarding using the information from the **Manual Port Mapping** window. See your router documentation for instructions on how to map ports. For information on configuring routers and mapping ports, visit **http://www.portforward.com/english/routers/port_forwarding/routerindex.htm**.

4. To check the status of each port through the firewall, click **Check Firewall**. The **Test Firewall** window appears.


Test Firewall...
Service | Status | Service | Status
Internet Access
Web Server (HTTP)
RTP Audio (Unit 1, Port 1)
RTP Audio (Unit 1, Port 2)
RTP Audio (Unit 1, Port 3)
RTP Audio (Unit 1, Port 4)
RTP Audio (Unit 2, Port 1)
RTP Audio (Unit 2, Port 2)
RTP Audio (Unit 2, Port 3)
RTP Audio (Unit 2, Port 4)
VoIP Signalling (SIP)
IP Phone Configuration (TFTP)
Details >>
Test Ports
Close

5. Select the services you want to check.

6. Click **Test Ports**. The system will check the ports for the selected services.

Configuring the router manually

If you cannot access the router configuration through the **IP Configuration** page, configure the router manually.

1. Open the router configuration and navigate to the screen used to set up port forwarding. See your router documentation.
2. Map port 5060 (Type: UDP) to the TalkSwitch unit acting as local proxy. Port 5060 is the default port for forwarding SIP signaling data to the TalkSwitch system.

   If required, you can map a different port. In this case, enter the port in the *SIP signalling port* box of the **VoIP Settings** window.
3. Map the following ports (Type: UDP) to the TalkSwitch units. These RTP ports forward voice traffic to the TalkSwitch system by default.

VoIP Lines/Unit	Unit 1	Unit 2	Unit 3	Unit 4
4	6000–6006	6010–6016	6020–6026	6030–6036
8	6100–6114	6200–6214	6300–6314	6400–6414

   If required, you can map different ports. In this case enter the first port in the *Starting RTP port* box of the **VoIP Settings** window.
4. If you are setting up external IP extensions, map port 69 (Type: UDP) to the TalkSwitch unit acting as local proxy. Port 69 allows the TalkSwitch system to configure external IP extensions.
5. If you are setting up TalkSwitch IP phones as external IP extensions, map port 8484 (Type: TCP) to the TalkSwitch unit acting as local proxy. Port 8484 allows the TalkSwitch system to configure TalkSwitch IP phones as external IP extensions.
6. Map port 9393 (Type: TCP) to the TalkSwitch unit acting as local proxy. Port 9393 allows remote configuration of the TalkSwitch system.
7. If available, enable Quality of Service (QoS) to give voice traffic priority over data.
8. Save the configuration to the router.

EXTERNAL IP EXTENSIONS

Warning: Using an external IP extension to call an emergency service number will not send the correct address to the emergency operator. We strongly recommend that you apply a warning label to any external IP extension stating:

If an emergency call is made from this phone, you must provide your address to the emergency operator.

An external IP extension is an IP phone located outside the office. It is configured as a local extension of a TalkSwitch system, but connects to the system over the Internet or private data network. A user can receive or place a call with their external IP extension through the TalkSwitch unit to the standard telephone network, or to a VoIP network. An external IP extension requires at least one VoIP-enabled TalkSwitch unit at the office.

For example, the following illustration shows external IP extension 151 at a home office, and 152 at a branch office. The external IP extensions connect to the TalkSwitch unit at the main office over the Internet. In this example, a user can receive or place a call with their external IP extension through the TalkSwitch unit to the standard telephone network.

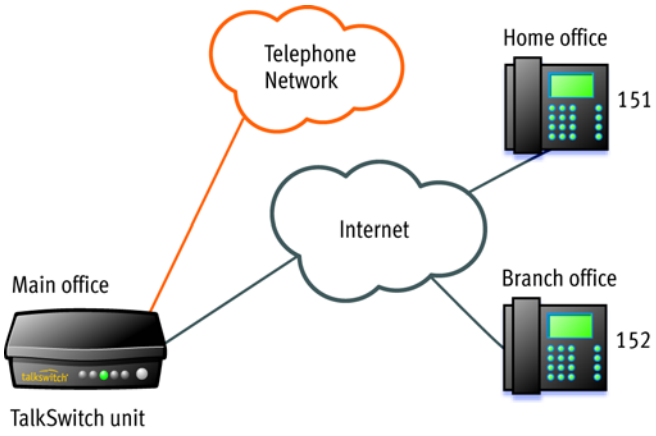


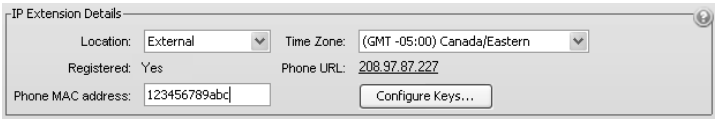
Figure 18: External IP extensions

Before proceeding, ensure you have:

- Connected the TalkSwitch system at each location to a network (page 231).
- Configured local and public IP addresses (page 232).
- Configured the router (page 235).

To set up an external IP extension, use the following steps:

1. Add the external IP extension, as described in *Adding IP phones* on page 81. Procedures are available for adding TalkSwitch, Polycom, Grandstream, Counterpath and selected other phones (depending on the market).
2. Verify operation of the external IP extension:
   a) Select the **Local Extension/Fax** page.
   b) Select the external IP extension.
   c) Confirm *Registered* is *Yes*, and the *Phone's IP address* appears in the *IP Extension Details* area.



3. Optionally, perform advanced VoIP configuration:
   a) All VoIP lines are shared by default. You can reserve VoIP lines for the external IP extensions. See *Reserving VoIP lines* on page 249.
   b) External IP extensions will use the preferred codec selected for the TalkSwitch profile. The default preferred codec is *G.729*. You can select the *Preferred codec* for the TalkSwitch profile. See *Setting codec options* on page 246.

TALKSWITCH VOIP NETWORK

A TalkSwitch VoIP network has at least one SIP device at each location. A SIP device is a VoIP-enabled TalkSwitch unit, IP telephone or SIP gateway. One TalkSwitch unit in the network acts as the SIP server. The other SIP devices are SIP clients.

A user, auto attendant or call cascade can dial a three-digit VoIP number (ranging from 250 to 299) to reach an extension, auto attendant or voice mailbox at any location.

For example, the following illustration shows a TalkSwitch VoIP network with two locations. Calls between the offices are routed over the Internet, and local calls are routed over the traditional telephone network.

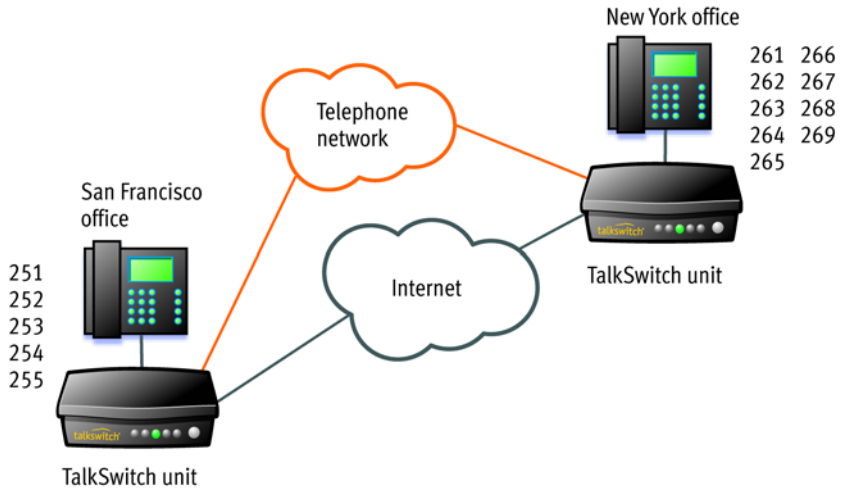


Figure 19: TalkSwitch VoIP network

Five VoIP numbers are assigned to San Francisco (251 to 255), and nine are assigned to New York (261 to 269). These VoIP numbers can be dialed directly from any extension, auto attendant or call cascade in the network.

The TalkSwitch unit in the San Francisco office has a static public IP address, so is designated as the SIP server. The TalkSwitch unit in New York is a SIP client.

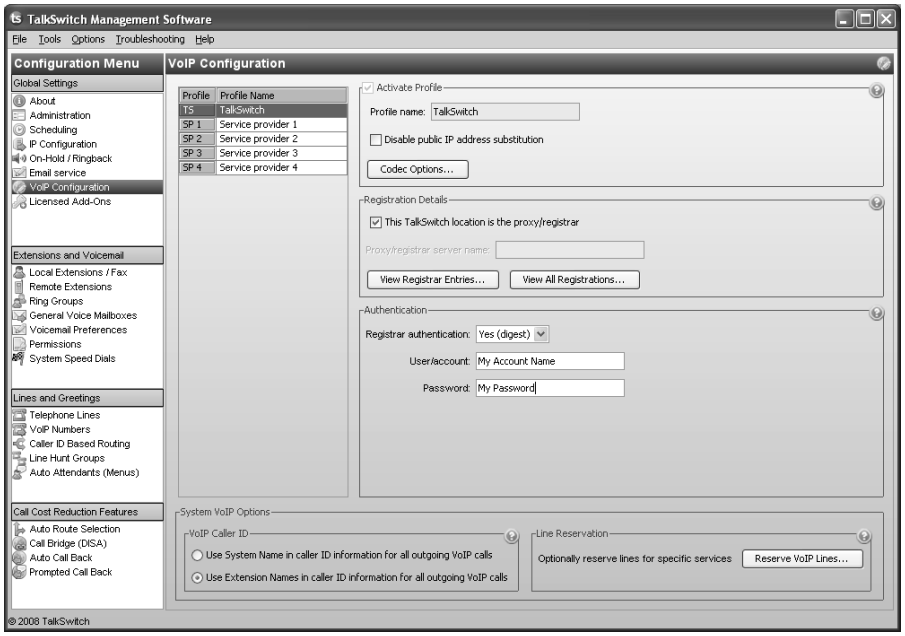
Before proceeding, ensure you have:

- Connected the TalkSwitch system at each location to a network (page 231).
- Configured local and public IP addresses (page 232).
- Configured the router (page 235).

SETTING UP A TALKSWITCH PROFILE FOR THE SIP SERVER

The TalkSwitch profile for the SIP server shows registration details and contains authentication information.

1. Start the TalkSwitch management software, and connect to the TalkSwitch system that will act as the SIP server.
2. Select the **VoIP Configuration** page.
3. Select **Profile TS**, which is for configuring the TalkSwitch profile.

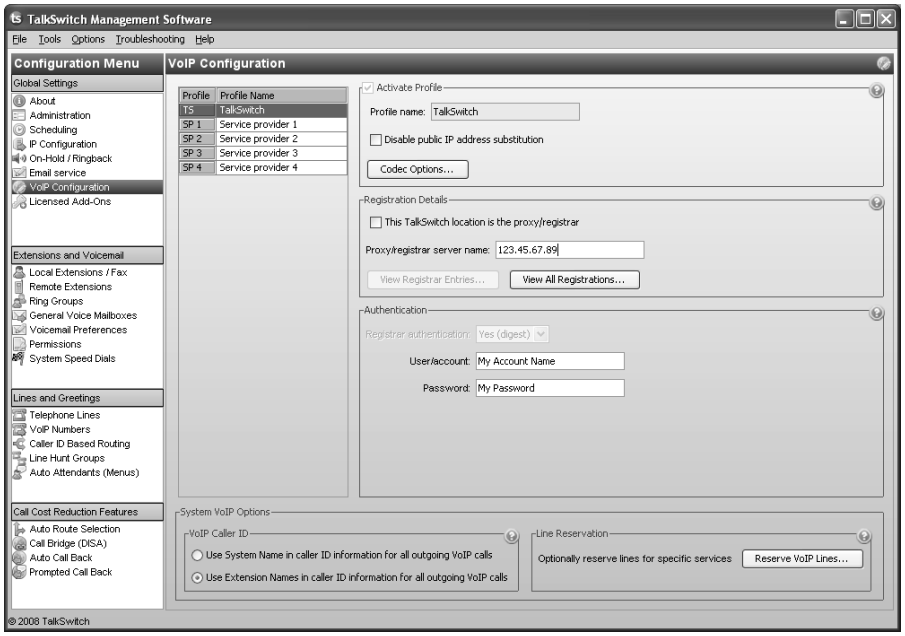


4. Select the *This TalkSwitch location is the proxy/registrar* check box.
5. Set *Registrar authentication* to *Yes (digest)*. This restricts access and allows only authorized devices to connect.
6. Enter a *User/account* name. All SIP clients must use this name.
7. Enter a *Password*. All SIP clients must use this password.
8. If you configured the router with non-default ports:
   a) Choose **Options > VoIP Trunking**. The **VoIP Settings** window appears.
   b) Set the *SIP Signalling Port* and/or *Starting RTP Port* as required.

SETTING UP A TALKSWITCH PROFILE FOR A SIP CLIENT

The TalkSwitch profile for a SIP client contains registration details and authentication information.

1. Start the TalkSwitch management software, and connect to the TalkSwitch system that will act as the SIP client.
2. Select the **VoIP Configuration** page.
3. Select **Profile TS**, which is for configuring the TalkSwitch profile.



4. Leave the *This TalkSwitch location is the proxy/registrar* check box cleared.
5. Enter the *Current public IP address* or *Fully qualified domain name* (FQDN) of the SIP server into the *Proxy/registrar server name* box.

   If the SIP server is using a non-default *SIP signalling port* (i.e. if the SIP server is not using 5060 as the *SIP signalling port*), specify the port number after a colon at the end of the public IP address or FQDN (e.g. 123.45.67.89:3456).
6. Enter the *User/account* name from the SIP server. All SIP clients must use this name.
7. Enter the *Password* from the SIP server. All SIP clients must use this password.

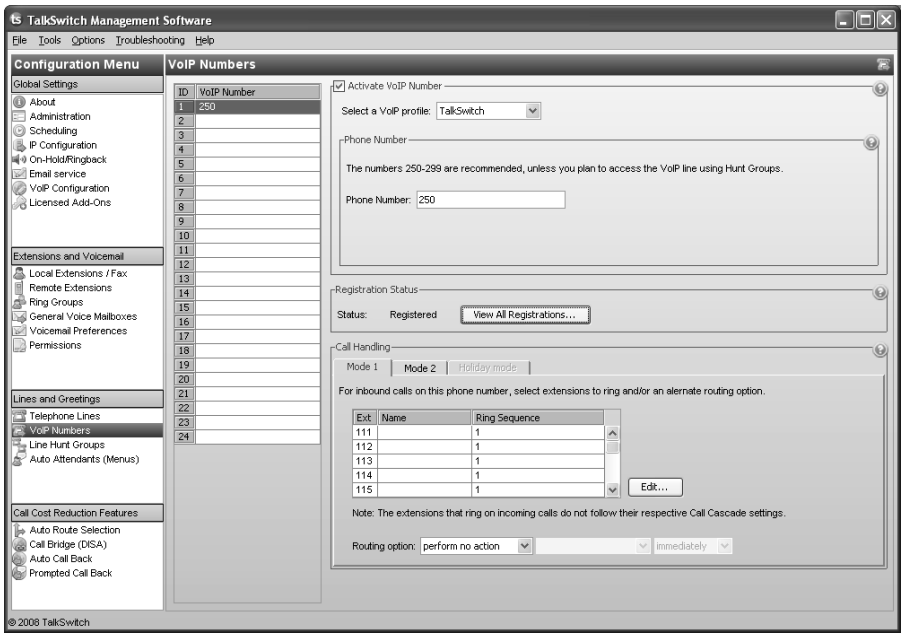
CONFIGURING VOIP NUMBERS FOR A TALKSWITCH VOIP NETWORK

A VoIP number is like a telephone number, and is used to dial a TalkSwitch system at a particular location. Each VoIP number must be unique (i.e. only used at one location).

VoIP numbers 250 to 299 permit direct dialing from any extension, auto attendant or call cascade to any extension, auto attendant or voice mailbox in the TalkSwitch VoIP network. No hunt group number is required. Longer VoIP numbers can also be used, but the user must first dial a hunt group number.

To assist in assigning VoIP numbers for your network, use the *TalkSwitch VoIP network administration form* on page 254.

1. Select the **VoIP Numbers** page.



2. Select a VoIP number slot.
3. Select the *Activate VoIP Number* check box.
4. Set the VoIP profile to *TalkSwitch*.
5. Enter the VoIP number into the *Phone number* box.
6. Set up call handling for the VoIP number. For more information, click the **Help** icon ( ) in the *Call Handling* area.
7. Repeat steps 2 to 6 for each additional VoIP number.

SETTING UP LINE HUNT GROUPS

IMPORTANT: Ensure that hunt group 9 or 0 is assigned to the group of telephone lines or VoIP trunks used for calls to emergency services.

If a VoIP-enabled TalkSwitch unit is present, hunt group 88 uses the TalkSwitch VoIP network by default, and the other hunt groups use telephone lines. You can modify these default settings as described on page 248.

Note that the user does not have to dial the hunt group number before VoIP numbers 250–299: these automatically use the TalkSwitch VoIP network.

If 50 VoIP numbers aren't enough for the network, you can set up longer VoIP numbers. To reach a longer VoIP number, the user will first dial a hunt group number.

A local extension can be restricted to a set of hunt groups in order to control access to VoIP networks. See *Hunt group permissions* on page 64.

ADVANCED VOIP CONFIGURATION

Optionally, perform advanced VoIP configuration.

1. The default preferred codec is *G.729*. You can select the *Preferred codec* for the TalkSwitch profile. All locations must use the same codec. See *Setting codec options* on page 246.
2. VoIP calls use extension names for caller ID by default. You can set the source for VoIP caller ID at each location. The same setting will be used by the TalkSwitch profile and all service provider profiles. See *Setting up caller ID* on page 249.
3. All VoIP lines are shared by default. You can reserve VoIP lines for the TalkSwitch profile at each location. See *Reserving VoIP lines* on page 249.

SAVING SETTINGS TO TALKSWITCH

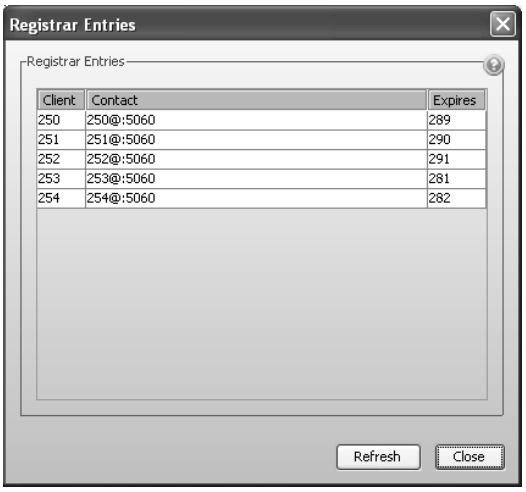
To transfer settings from your computer to the TalkSwitch system, choose **File > Save**. A window appears indicating the configuration is being sent.

VERIFYING THE TALKSWITCH VOIP NETWORK

If you are configuring the TalkSwitch profile for the SIP server, the window enables the **View Registrar Entries** button. Clicking the button shows the **Registrar Entries** window with a list of VoIP numbers, their IP addresses and port numbers, and the number of seconds until their registrations with the SIP server will expire.

1. Select the **VoIP Configuration** page.

2. Select **Profile TS**, and then click **View Registrar Entries**. The **Registrar Entries** window appears.



SERVICE PROVIDER VOIP NETWORK

Using a service provider VoIP network requires at least one VoIP-enabled TalkSwitch unit. The service provider acts as the SIP server. The service provider assigns the VoIP numbers and VoIP configuration parameters.

For example, the following illustration shows a service provider VoIP network.

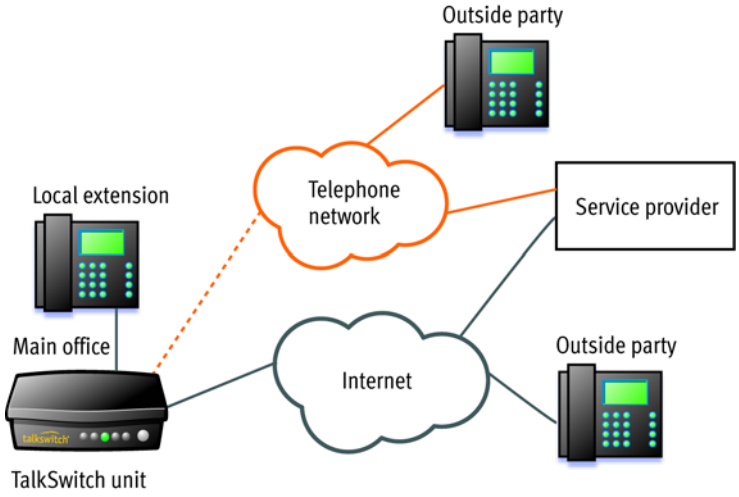


Figure 20: Service provider VoIP network

To reach an outside party, the user dials the hunt group number associated with the service provider profile, and then dials the phone number. The TalkSwitch unit routes the call over the Internet to the service provider. If the outside party doesn't use the same service provider, they are connected over the standard telephone network. If the outside party does use the same service provider, they are connected over the Internet.

To reach a local extension, an outside party dials the VoIP number that was assigned to the TalkSwitch system by the service provider. The call is routed through the telephone network to the service provider. The service provider routes the call over the Internet to the TalkSwitch system, which directs the call to the local extension.

Before proceeding, ensure you have:

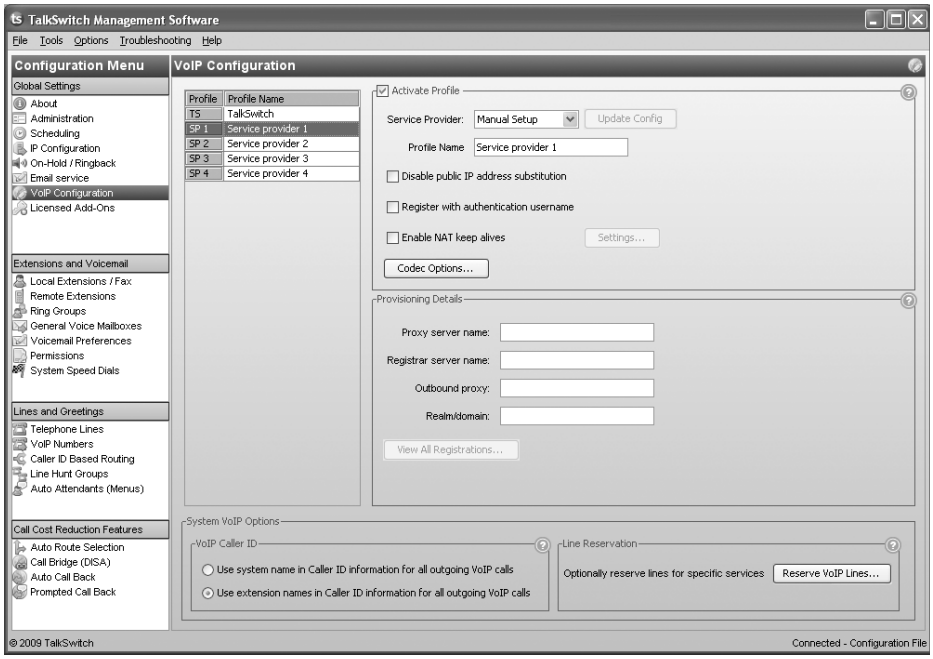
- Connected the TalkSwitch system at each location to a network (page 231).
- Configured local IP addresses (page 232), and public IP addresses if required.
- Configured the router (page 235).

SETTING UP A SERVICE PROVIDER PROFILE

If you are using a TalkSwitch-certified VoIP service provider, visit the support section at **http://global.talkswitch.com** to access the service configuration guide for your VoIP service provider. Otherwise use the procedures in this section.

A service provider profile contains the registration details for connecting to the service provider's SIP server.

1. Start the TalkSwitch management software.
2. Select the **VoIP Configuration** page.
3. Select a **Profile** (**SP 1** to **SP 4**), which are for configuring service provider profiles.



4. Select the *Activate Profile* check box.
5. Enter the *Profile name*. The default profile name is *Service provider n* (e.g. *Service provider 1*).
6. If your service provider requires you to register using your private IP address, select the *Disable public IP address substitution* check box.
7. If your service provider requires you to set up keep alive messages, and if your router does not support uPNP, set up keep alive messages.

   Keep alive messages maintain the connection from the TalkSwitch unit, through your router, to the service provider's SIP server.

   a) Select the *Enable NAT keep alives* check box.

b) Click **Settings**. The **Keep Alive Settings** window appears.

c) Select the type of ping, as directed by your service provider. Choices are:

- *Simple ping* — A standard ping message that works with all SIP servers.
- *Nortel ping* — A longer but more reliable ping message that only works with specific Nortel SIP servers (e.g. Nortel MCS 5200).

d) If necessary, change the ping frequency, as directed by your service provider. The default setting is 45 seconds.

8. Enter the IP addresses or FQDNs, as provided by the service provider, into the following boxes. If the service provider does not specify a value, leave the box blank.

- *Proxy server name*
- *Registrar server name*
- *Outbound proxy*
- *Realm/domain*

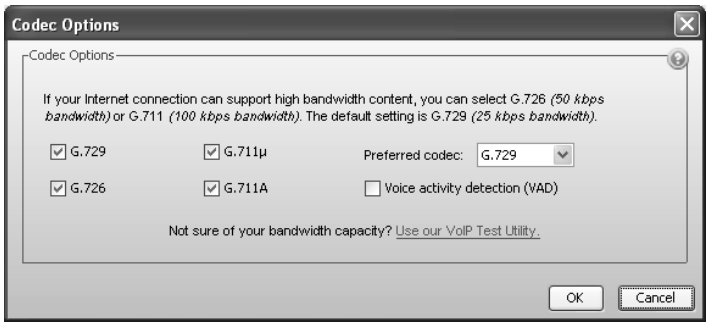
SETTING CODEC OPTIONS

If you are using a TalkSwitch-certified VoIP service provider, visit the support section at **http://global.talkswitch.com** to access the service configuration guide for your VoIP service provider. The service configuration guide lists the supported codecs.

A codec is a method of compressing and decompressing audio signals for communication across a network. TalkSwitch supports the G.729, G.726 and G.711 (µ-law or A-law) codecs for VoIP calls. If your service provider or equipment requires specific codecs for VoIP or Fax over IP calls, you can restrict TalkSwitch to use the required codec.

The **Codec Options** window allows you to select the codecs your system can use, specify the preferred codec, and clear the unsupported codecs. You can specify the codecs for the TalkSwitch profile, and for each service provider profile. External IP extensions will use the preferred codec specified in the TalkSwitch profile.

1. Select the **VoIP Configuration** page.
2. Click **Codec Options**. The **Codec Options** window appears.


Codec Options
Codec Options
If your Internet connection can support high bandwidth content, you can select G.726 (50 kbps bandwidth) or G.711 (100 kbps bandwidth). The default setting is G.729 (25 kbps bandwidth).
G.729
G.711µ
Preferred codec:
G.729
G.726
G.711A
Voice activity detection (VAD)
Not sure of your bandwidth capacity? Use our VoIP Test Utility.
OK
Cancel

3. Select the codecs your system can use, and clear the unsupported codecs. The following codecs are supported:

   - *G.729* — This codec provides good quality. It requires the least bandwidth and accommodates the highest number of concurrent calls.
   - *G.726* — This codec provides better quality. It requires more bandwidth and accommodates fewer concurrent calls.
   - *G.711*µ and *G.711A* — This codec provides high quality and supports Fax over IP. It requires the most bandwidth and accommodates the fewest number of concurrent calls. G.711µ is used in North America and Japan. G.711A is used elsewhere.

4. Select the *Preferred codec*. *G.729* is the default codec.

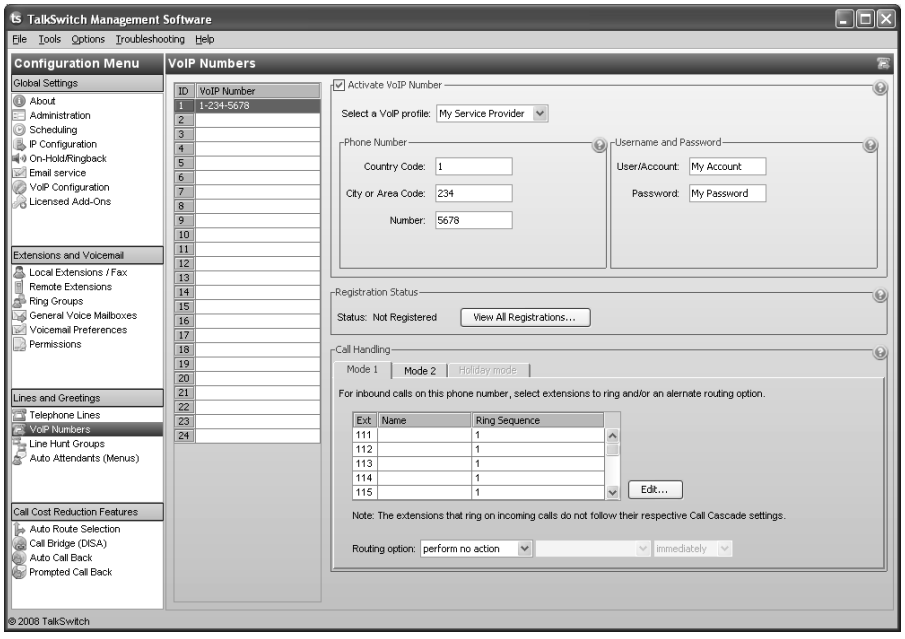
5. Set *Voice activity detection (VAD)*. Enabling VAD reduces voice bandwidth when no speech is detected, and reduces transmission of background noise. We recommend disabling VAD to keep bandwidth available for speech.

CONFIGURING VOIP NUMBERS FOR A SERVICE PROVIDER VOIP NETWORK

If you are using a TalkSwitch-certified VoIP service provider, visit the support section at **http://global.talkswitch.com** to access the service configuration guide for your VoIP service provider. Otherwise use the procedures in this section.

A VoIP number is like a telephone number, and is used to dial a TalkSwitch system at a particular location. Your service provider will assign VoIP numbers for each location.

1. Select the **VoIP Numbers** page.



2. Select a VoIP number slot.

3. Select the *Activate VoIP Number* check box.

4. Set the VoIP profile to the service provider (e.g. *My Service Provider*).

5. Enter the *Country Code* (if required), the *City or Area Code* and the *Number*.

6. Enter the *User/account* and the *Password* (if required) for this number.

7. Set up call handling for the VoIP number. For more information, click the **Help** icon ( ) in the *Call Handling* area.

8. Repeat steps 2 to 7 for each additional VoIP number.

SETTING UP LINE HUNT GROUPS

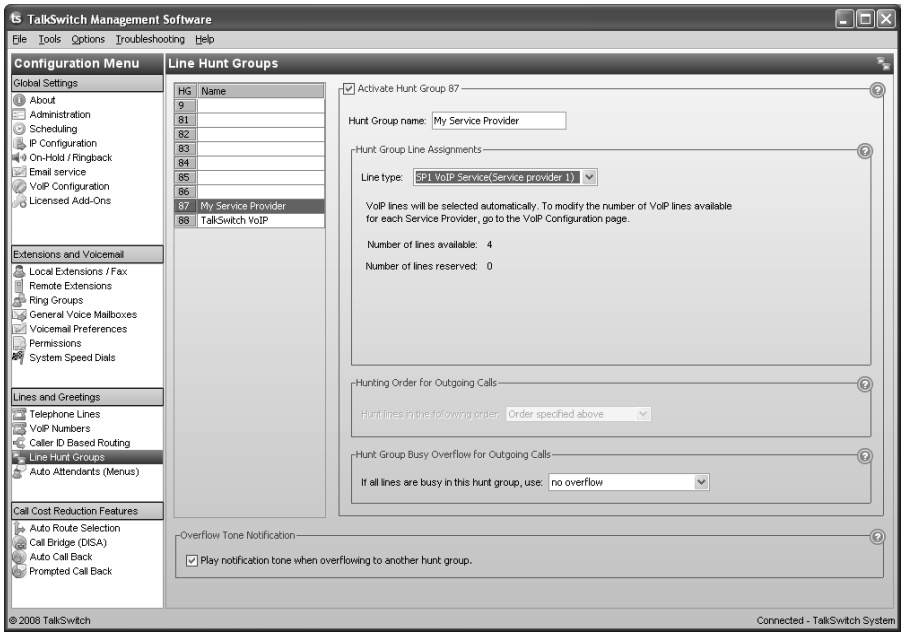
IMPORTANT: Ensure that hunt group 9 or 0 is assigned to the group of telephone lines or VoIP trunks used for calls to emergency services.

A line hunt group is a set of lines that are available for making an outbound call. It can use selected telephone lines, all VoIP lines associated with the TalkSwitch VoIP network, or all VoIP lines associated with a service provider VoIP network.

The configuration has nine different hunt groups. If a VoIP-enabled TalkSwitch unit is present, hunt group 88 uses the TalkSwitch VoIP network by default, and the other hunt groups use telephone lines. You can modify these default settings as required. If you are using multiple service provider VoIP networks, set up a hunt group for each service provider.

A local extension can be restricted to a set of hunt groups in order to control access to VoIP networks. See *Hunt group permissions* on page 64.

1. Select the **Line Hunt Groups** page.



2. Select a hunt group slot.

3. If necessary, select *Activate Hunt Group*. All hunt groups are active by default.

4. Enter a *Hunt Group name*.

5. Set *Line type* to *TalkSwitch VoIP* or *SPn VoIP Service*.

6. Select the overflow hunt group. Choices are *no overflow* and the other activated hunt groups. If you select an overflow hunt group, ensure it contains different lines than the original hunt group.

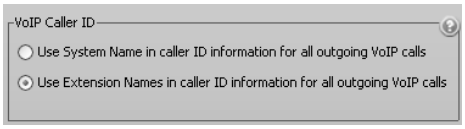
ADVANCED VOIP CONFIGURATION

Optionally, perform advanced VoIP configuration.

Setting up caller ID

The *VoIP Caller ID* area allows you to set up the source for caller ID for outbound VoIP calls. The same setting is used for the TalkSwitch profile and all service provider profiles. Extension names are used by default.

1. Select the **VoIP Configuration** page.
2. Set the caller ID for outbound VoIP calls.
   - To use the *System name* from the **Administration** page, select *Use System Name in caller ID information for all outgoing VoIP calls*.
   - To use the *First name* and *Last name* from the **Local Extension/Fax** page set up for each extension, select *Use Extension Names in caller ID information for all outgoing VoIP calls*.

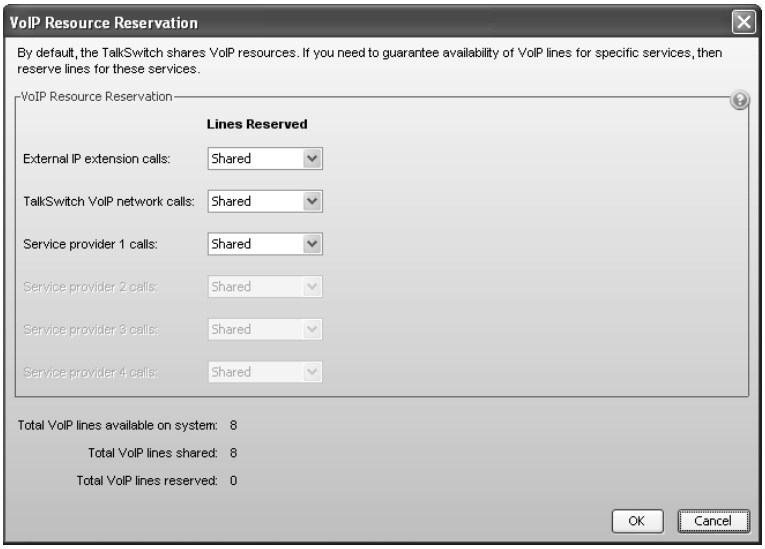

VoIP Caller ID
Use System Name in caller ID information for all outgoing VoIP calls
Use Extension Names in caller ID information for all outgoing VoIP calls

Reserving VoIP lines

By default, all TalkSwitch VoIP lines are available for external IP extensions, TalkSwitch VoIP network calls, and/or service provider calls on a first-come first-served basis. You can also reserve VoIP lines for a specific use. For example, you can set aside two lines for external IP extensions.

Reserving VoIP lines guarantees resources for a specific network. Sharing VoIP lines uses them most efficiently.

1. Select the **VoIP Configuration** page.
2. Click **Reserve VoIP Lines**. The **VoIP Resource Reservation** window appears.


VoIP Resource Reservation
By default, the TalkSwitch shares VoIP resources. If you need to guarantee availability of VoIP lines for specific services, then reserve lines for these services.
VoIP Resource Reservation
Lines Reserved
External IP extension calls: Shared
TalkSwitch VoIP network calls: Shared
Service provider 1 calls: Shared
Service provider 2 calls: Shared
Service provider 3 calls: Shared
Service provider 4 calls: Shared
Total VoIP lines available on system: 8
Total VoIP lines shared: 8
Total VoIP lines reserved: 0
OK
Cancel

3. Select the number of VoIP lines to set aside for each service. Choices range from 1 to the number of unreserved VoIP lines at this location. *Shared* means no VoIP lines are reserved for this service. Only unreserved lines will be available.

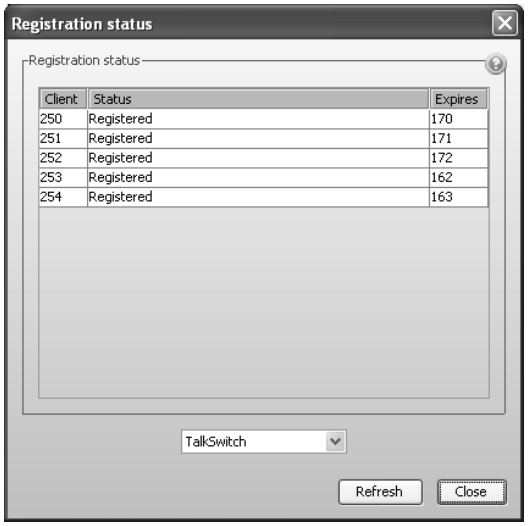
SAVING SETTINGS TO TALKSWITCH

To transfer settings from your computer to the TalkSwitch system, choose **File > Save**. A window appears indicating the configuration is being sent.

VERIFYING REGISTRATION

Clicking the **View All Registrations** button shows a window with a list of VoIP numbers, their registration status, and the number of seconds until their registrations with the SIP server will expire. This confirms the TalkSwitch system is registered with a SIP server.

1. Select the **VoIP Configuration** page.
2. Click **View All Registrations**. The **Registration status** window appears.
3. Choose *All Registered Numbers* or an active profile.



FAQ

How does VoIP work?

Voice over IP (VoIP) sends calls over the Internet or any Internet Protocol (IP) network by converting voice traffic into data packets. These packets are routed over the IP network the same way as other data, reassembled at the point of arrival, and then converted back to voice.

Does a VoIP call sound like an analog phone call?

Under the right conditions, VoIP calls sound as good or better than analog phone calls. Voice quality is affected by numerous factors, including how it is compressed and decompressed for delivery over the Internet and how packets are processed. A few steps can help ensure call quality.

- **Use high-speed connections on both ends of the call**
  VoIP requires a broadband connection to support simultaneous calls. Please visit the support section of our site at **http://global.talkswitch.com** to determine what your Internet connection can support. Good quality VoIP is not possible over a dial-up modem connection.

- **Choose the right codec for your location**

  - The default codec is **G.729** (8 Kbps), using approximately 25 Kbps bandwidth upstream and 25 Kbps bandwidth downstream for each call. G.729 provides very good call quality while minimizing bandwidth usage.

  - The **G.726** (32 Kbps) codec is a better quality solution compared to the G.729 codec. However, it requires more bandwidth per call. A G.726 call typically requires 50 Kbps bandwidth upstream and 50 Kbps bandwidth downstream for each call.

  - The **G.711** (64 Kbps) codec provides the best voice quality. The trade-off is the bandwidth requirement. G.711 calls typically require up to 100 Kbps bandwidth upstream and 100 Kbps bandwidth downstream.

What happens if the power goes out or if the IP network to VoIP fails?

To ensure a reliable network connection, all elements of the VoIP network should be connected to back-up power supplies (UPS). These elements should include LAN switches, routers, firewalls, broadband connection devices (i.e. cable modems, DSL modems), and VoIP devices. If the power goes out at the Internet Service Provider, no VoIP calls can be made. Calls can still be placed over the telephone lines.

Can a firewall prevent VoIP calls from passing through?

The purpose of a firewall is to control what kind of traffic enters and leaves your network. The TalkSwitch unit is designed to allow VoIP calls to traverse firewalls properly. To allow VoIP calls to pass through your firewall, you may need to use the port forwarding feature on your firewall.

What is SIP?

The Session Initiation Protocol (SIP) is a signalling protocol used for establishing sessions in an IP network. A session could be a simple 2-way telephone call or it could be a collaborative multi-media conference session.

Over the last couple of years, the Voice over IP community has adopted SIP as its protocol of choice for signalling. SIP is an RFC standard (RFC 3261) from the Internet Engineering Task Force (IETF).

SIP initiates, terminates and modifies sessions.

What is a VPN?

VPN stands for Virtual Private Network. It is a means of having the advantages of a private network on a shared public infrastructure like the Internet.

Can a VPN help to carry data securely over the Internet?

A VPN provides security by using encryption/decryption. Using a process called tunneling, private data is encrypted and encapsulated before being sent across the network. The packets of information are sent to their destination. Only the ones with the correct protocol information can enter a company's network.

A VPN is most useful for providing employees with remote access to the company network without compromising security. Encryption can have a negative impact on the call quality because of the increased overhead on the network connection. IPSec adds approximately 10% overhead to VPN traffic.

What's the difference between a public IP address and a private IP address?

A public IP address is a globally unique number that identifies a device on the Internet. If you want someone on the Internet to connect to you, they must know your public address. This is also known as your real or external address.

A private IP address is typically assigned to a device on a Local Area Network (LAN) and cannot be routed outside the LAN. Private IP addresses are usually in the range 192.168.x.x, 172.16.x.x or 10.x.x.x. These IP addresses are used where you have multiple computers all sharing the same Internet connection.

To access the Internet, a computer or VoIP device must have an IP address. So what do you do if your ISP (Internet Service Provider) has provided you with only one IP address but you have more than one device that requires Internet access? The solution is to give each of the devices on the LAN a private IP address. A router makes them work by performing Network Address Translation (NAT — See the next topic for details).

Addresses of the form 192.168.x.x, 172.16.x.x or 10.x.x.x are not recognized on the Internet and can only be used for private networks. For example, if you tell someone on the Internet to connect to you using a 192.168.x.x address, it will not work. Instead, you must provide your real/public IP address.

What is NAT and how does it affect VoIP?

Network Address Translation (NAT) allows multiple devices to share the same external IP address to access resources on the Internet. NAT is typically used to allow all the devices in a subscriber's local area network to access the Internet through a router with a single public IP address assigned by the Internet Service Provider.

If a VoIP device is sitting behind a router that performs NAT, the private IP address assigned to the VoIP device is not usable for communications with the entities outside the private network. The VoIP device must substitute the private IP address information with the proper external IP address/port in the mapping chosen by the underlying NAT to communicate with a particular public peer address/port. TalkSwitch can automatically check for the current public IP address and substitute the private IP address with the public address so that VoIP traffic is properly routed through NAT.

What's the difference between a static IP address and a dynamic IP address and what effect does this have on VoIP?

A static IP address is an address that is permanently assigned to a device. Typically, a static IP address has to be assigned by the network administrator or Internet Service Provider (ISP).

A dynamic IP address is an address that is temporarily assigned to devices by a DHCP or PPP server that maintains and assigns a pool of IP addresses.

When deploying VoIP devices, it's best to have a static public IP address. This prevents destinations from becoming unreachable for periods of time when the IP address changes. But since this can be an expensive option, TalkSwitch supports the ability to handle dynamic public IP addresses for the Internet connection. TalkSwitch automatically checks for the current public IP address and substitutes the private IP address with the public address, so that VoIP traffic is properly routed through NAT.

What is a SIP server?

The basic IP architecture is client/server in nature. The main entities in the SIP server are the proxy, redirect server and the registrar.

Every TalkSwitch unit has a fully integrated SIP server for easy deployment and maintenance of your VoIP network. If you are deploying a multi-site configuration, one TalkSwitch unit will act as the SIP server for the entire VoIP network. The other IP end-points will register with the SIP server and all calls will be negotiated through the SIP server.

When you first set up your VoIP device (TalkSwitch unit, gateway or phone), it will register itself with the SIP server so calls can be directed to your location.

When you initiate a VoIP call, your VoIP device will contact the SIP server in order to obtain the contact information for the destination you are trying to reach. Once the contact information is received, the call is established directly between your location and the destination. The SIP server is no longer involved in the call.

If the location of the SIP server has a dynamic IP address, you have to use the dynamic DNS since the location of the SIP server becomes unreachable as soon as the IP address changes.

What is dynamic DNS?

Dynamic DNS is a Domain Name Service used with dynamic IP addresses. DNS is used to allow devices to find other devices on the Internet by domain name rather than by IP address. Dynamic DNS allows you to use a Fully Qualified Domain Name (FQDN) to locate a device with a dynamic IP address. The FQDN does not change. There are companies on the Internet who can provide you with a domain name that will work even if your IP address changes. Such companies are **www.dyndns.org**, **hn.org** or **www.no-ip.com**. Please consult with them how to use their service. The basic idea is you will receive a domain name like myname.homeip.net, which will continue to point to your IP address as it changes. This is important for the location of the SIP server if the public IP address is dynamic. All other locations will point to the SIP server's domain name instead of its IP address.

What is the drawback of using dynamic DNS?

If the IP address of the location of the SIP server changes, but the new IP address hasn't been propagated, other devices will not be able to contact the SIP server in order to place VoIP calls.

When DNS records are updated, they have to be sent all over the world so everyone will be aware of the changes. Basically, one DNS server connects to another and swaps information. In many cases, the DNS servers are updated very quickly. Updates should not take longer than 15 minutes to propagate throughout the DNS system. It is possible propagation could take longer due to unforeseen traffic but it shouldn't take much more than an hour at the most.

How often will my public IP address change?

This depends on the policies of your Internet Service Provider (ISP, the company that provides your Internet access). It can be a matter of days, weeks or even months before your IP address changes. A power loss or reboot to your cable or DSL modem usually results in the assignment of a new IP address.

TalkSwitch has technology that helps keep this information up to date so the system can function properly after an IP address change has been detected and updated.

What is a port number?

In an IP address like 64.26.209.103:5060, the port number is 5060; it is the number after the colon.

Port numbers are required so a computer or device can support multiple applications or streams of simultaneous data communication. Each stream or application uses a unique port number. You can think of it like mailing a letter to a friend in an apartment building at a specific address. You need to specify the apartment number as well as the address of the building, so the mail can reach the correct final destination.

TALKSWITCH VOIP NETWORK ADMINISTRATION FORM

Use this form to keep track of VoIP numbers assigned to locations within a TalkSwitch VoIP network.

Location of the SIP server:_____

Equipment at the SIP server location:_____

VoIP numbers assigned to the SIP server location:_____

VoIP numbers assigned to locations

Location name	Equipment	VoIP numbers

CHAPTER 6

TROUBLESHOOTING AND SUPPORT

TROUBLESHOOTING

Solutions to the following problems are described in this chapter:

- Configuration
- Auto attendant
- Music on hold
- Call routing
- Answering and fax machines
- Local extensions
- Multiple TalkSwitch units
- VoIP

Some problems might be due to physical connections such as loose cables. Before you start to analyze possible configuration issues. See *Verifying the connections* on page 21.

Configuration

I am not able to retrieve settings from TalkSwitch

1. Reboot TalkSwitch. On the TalkSwitch front panel, turn the power button off and back on.
2. Make sure you do not have any communications programs running at the same time you are using the TalkSwitch management software. These can include Palm Pilot, Hot Sync, TalkWorks, digital camera software or others. These programs tend to occupy COM ports, making them unavailable for other programs.

I am unable to configure TalkSwitch using a touchtone phone

1. You cannot configure TalkSwitch using a phone sharing the same line in parallel. You have to configure TalkSwitch using one of the extensions plugged into the back panel of TalkSwitch.
2. You cannot configure TalkSwitch using touchtone keypad commands while the TalkSwitch management software is open. If the software is closed and you hear the prompt *"I'm sorry, TalkSwitch is currently being configured"*, reboot TalkSwitch by turning the power off and on. Wait until the lights stop flashing and the **Data** light is on solid.
3. If you are using a local extension, check if the extension has direct line access to your telephone lines. If direct line access is enabled, press **Flash** or **Recall** on an analog extension or ** on a IP extension to receive an internal dial tone. Press **#** to enter command mode. Enter the correct password, if you have password protection.

4. From an out-of-office phone, call into TalkSwitch and wait for the auto attendant. Enter command mode by pressing **#** on an analog extension phone, or ***55#** on a TalkSwitch IP phone (note: other brands may use ***55 Send** or ***55 Dial**). Enter the correct password, if you have password protection.

Auto attendant

The auto attendant does not play when calls come in

1. Check if the line light on an incoming call is flickering while the phone is ringing. If not, replace the phone cord.
2. Make sure you have an auto attendant message recorded.
3. Ensure the system is running the correct mode. Open the TalkSwitch management software. Select **Scheduling**. Check the *Current mode* in the *Modes* area.
4. Ensure the telephone line has the correct auto attendant setup. Select **Telephone Lines**. Ensure the right auto attendant is selected for the mode, telephone number, and telephone line. Try setting the auto attendant to answer *Immediately*.

The auto attendant message is broken up or very faint

1. Open the TalkSwitch management software. Choose **Troubleshooting > Telephone Lines > Audio**. Adjust the *Transmit levels* volume setting.
2. The quality of the microphone in the telephone handset you used to record your auto attendant can affect the quality of your recording. Try recording the auto attendant from another extension/telephone.

The auto attendant answers calls before any of the extensions ring

If TalkSwitch users wish to have their local extensions ring before the auto attendant picks up incoming calls, do the following:

1. Open the TalkSwitch management software. Select **Telephone Lines.** Select the telephone line, telephone number, and mode. Click **Edit**. Increase the number of rings.

The auto attendant is transferring calls to the wrong extension

1. Check if the extension jacks of your devices (phones, faxes) are plugged into the corresponding jacks on the TalkSwitch back panel (i.e. extension 111 is plugged into E1).

The auto attendant answers calls, but does not transfer them to the extensions

1. Make sure your extensions are plugged into the extension jacks on the back panel of TalkSwitch (instead of your telephone wall jacks).
2. Open the TalkSwitch management software. Expand **Auto Attendants (Menus)**. In the *Actions During Auto Attendant Playback* area, check if the actions and resources point to the correct extensions.

Calls are going to my phone company voicemail instead of being answered by TalkSwitch

This may be a result of two different situations:

1. If the line is currently busy and you do not have the hunt/rollover service from the phone company, then the call will go into voicemail.

2. If calls are ringing, then going into voicemail, TalkSwitch may be set to answer after a predetermined number of rings that is larger than the number of rings set for your phone company voicemail.

   To remedy this situation, decrease the number of rings before the auto attendant answers (see below) or increase the number of rings before your phone company voicemail answers.

To decrease the number of rings before an auto attendant answers:

1. Open the TalkSwitch management software, select **Lines and Greetings > Telephone Lines**. The *Telephone Lines* section appears.

Under the *Call Handling* section, set the *Routing option* to less than the phone company's voicemail number of rings, or to *immediately*.

How do I access my voicemail remotely?

To access your voicemail remotely:

1. Call into TalkSwitch. The auto attendant answers.
2. Press ******, followed by your extension number. This will take you into the standard voicemail options and allow you to listen to your messages and change voicemail options.

NOTE: If your inbound call is answered by a local extension, they can transfer you to your voicemail box by transferring your call by pressing ******, followed by your extension number.

Music on hold

Callers hear the “One moment please” message and then total silence when the auto attendant transfers their call to an extension

1. There is a problem with the music source. Check all connections and the power.

Callers hear only silence when put on hold at an extension

1. Make sure you use the **Flash** or **Recall** or **Link** button on an analog extension, or the **Hold** button on an IP extension, to put callers on hold. If you use the **Hold** button on an analog extension, callers will be placed on hold at the extension itself and not through TalkSwitch.
2. You have enabled the music on hold feature, but have not attached an audio device to the music jack on TalkSwitch.
3. You have enabled play the *Play music from file* feature but have not loaded a .wav music file to TalkSwitch.

Call routing

Callers are disconnected when transferring calls from one extension to another

1. If you are using the hook switch rather than the **Flash** or **Recall** button to put callers on hold, you may be holding the hook switch down too long and disconnect.

I can’t use my conference/3-way calling feature from the phone company

1. You need to make sure you have checked the box indicating you have 3-Way Calling/ Conference service or the Transfer and Clear service on the appropriate lines in the **Telephone Lines** page.

I'm unable to place intercom calls from a local extension

1. If the local extension has been set up for direct line access to your telephone lines, you need to press **Flash** or **Recall** on an analog extension, or ** on an IP extension, before intercom calling.

Answering and fax machines

The answering machine and/or fax machine picks up calls before they can be answered by telephone extensions

1. Check the ring sequence for the telephone line, telephone number and mode. Ensure the extension for your fax machine and/or answering machine is not set to answer calls at the same time as your telephone extensions. If necessary, remove the extension for the fax machine and/or answering machine from the list of extensions to ring on an incoming call.

Incoming faxes are not automatically detected and routed to the fax machine

1. Ensure you selected the correct extension in the *If a fax call is detected* list in the **Auto Attendants (Menus)** page.
2. Not all fax machines emit a CNG tone that TalkSwitch can detect and route. Therefore, you may consider incorporating an additional method of fax routing. This could include the use of a dedicated fax line or a distinctive ring number.
3. Many fax machines/modems will time-out or disconnect after 25 seconds on average if they have not connected with another fax machine/modem. If you set up the *Action Performed After Auto Attendant Playback* area, the fax call may time-out before it reaches your fax extension. If you have a long auto attendant message, you may want to shorten it or use another fax routing option.
4. The level of the auto attendant is too loud and is overpowering the CNG tone, affecting the detection. Try lowering the volume level of the auto attendant.

Local extensions

I am unable to access TalkSwitch lines with a local extension

1. Check your direct line access setting in the **Additional Settings** window of the **Local Extensions/Fax** page. If a local extension has not been given direct line access to your telephone lines, you must dial **9** (**0** in some regions) or **81**–**88** to access a line.
2. Check if *Activate automatic route selection* is enabled in the **Auto Route Selection** page. ARS can re-direct or block calls.

My PC (internal or external) modem cannot dial out

1. If direct line access to telephone lines is not enabled on the extension where your modem is connected, you will have to include a line hunt group prefix — **9** (**0** in some regions) or **81**–**88** before dialing the modem number.

One of the local extensions (telephones) does not ring

1. Some telephones require more voltage in order to ring. These phones are usually the older 'Bell' phones with mechanical ringers. These phones should not be used as TalkSwitch extensions.
2. Make sure the telephone cord you are using between the phone and TalkSwitch is working properly.
3. Ensure the ringer is turned on.

Extension(s) ring, but there is no caller

This can occur when a caller hangs up after the auto attendant has begun to play. In some areas, the telephone company's disconnect/clear signal is weak. The auto attendant may not be able to pick up the disconnect signal as soon as a caller hangs up. The auto attendant will not receive a response after playing its message and the call will fall through to your settings in the *Action Performed After Auto Attendant Playback* area in the **Auto Attendants** page.

When I try to access voicemail, I hear a busy tone

1. Check your direct line access settings in the **Additional Settings** window of the **Local Extensions/Fax** page in the TalkSwitch management software. If a local extension has been given direct line access to your telephone lines, you must first press **Flash** or **Recall** on an analog extension, or ****** on an IP extension, before dialing the mailbox. The same applies for dialing other extensions to initiate intercom calls.

Multiple TalkSwitch units

I can't reach an extension across the LAN

1. Make sure the other unit is connected to the LAN and powered on.
2. Pick up a local extension belonging to the other unit. If you hear a dial tone, check your LAN connections. On the rear panel, the top LED of the three LEDs next to the LAN port should be on. If there is no dial tone, make sure the unit is powered on.

The configuration does not show the other TalkSwitch unit(s)

1. Check your LAN connections. Are you able to communicate with another extension on the other unit?
2. Check the unit ID of each unit. If the lights on one or more units are flashing, check *What the flashing lights mean* on page 2 to determine the cause.

   If there is a conflict with the unit ID numbers assigned to each TalkSwitch unit, give each unit a unique unit ID. See *Setting or changing the unit ID* on page 17.

After recording an auto attendant greeting, I can't play it back

1. Try again in a few minutes. After recording an auto attendant on one of the TalkSwitch units, it will copy it to other TalkSwitch units on the LAN. During this process, you cannot listen to that particular auto attendant.

Calls are not always reaching my voicemail

When the system is in high use, all channels across the LAN might be in use, so calls cannot be transferred to extensions or voicemail. If this situation happens frequently, consider optimizing the way the system uses the lines.

1. In the **Line Hunt Groups** page, select the *For all hunt groups, hunt lines on same unit first* check box. This arrangement keeps most of the traffic on unit 1 and any outbound calls made from any extension will go out on the same unit the extension is connected to. This also applies for inbound calls that are forwarded to remote extensions. The call will be forwarded out on a line belonging to the same unit the call came in on. The call will only be forwarded on the other TalkSwitch unit if no telephone line on that first unit is available.
2. Ensure frequently used extensions are connected to units with enough lines out to reduce LAN traffic.

Example: You have two TalkSwitch units, six telephone lines and 12 employees. The six lines have the hunt/rollover service from the phone company. Suggested setup: Plug the four highest traffic lines into unit 1. Also, connect the eight highest phone usage employee extensions to unit 1. Connect the remaining lines and extensions to unit 2.

Calls across the LAN are lower in volume

1. Check the **Telephone Lines - Audio** page. You may need to change the transmit level or impedance matching on one or more lines to compensate for high loss lines. If these controls are set incorrectly and your lines are problematic, you may hear an echo when calls are made across the LAN.

VoIP

I hear a re-order tone when I try to dial a VoIP location

1. Confirm the number dialed is actually part of the VoIP group (i.e. registered with the same proxy/registrar server as your TalkSwitch, IP gateway or IP extension is registered with.
2. Ensure TalkSwitch has all proper settings in the **IP Configuration** and **VoIP Configuration** pages.
3. Ensure the number you are dialing is registered with your TS proxy registrar. Go to the **VoIP Configuration** page and click the **View Registrar Entries** button.

Callers complain the sound is distorted or choppy

1. Your broadband connection may not have enough upstream bandwidth to support many simultaneous VoIP calls. In the **Codec Options** window, you may need to disable the G.711 and G.726 codecs and only use G.729.
2. You may have too much data traffic on your Internet connection at the same time you are trying to make voice calls. We suggest you use a router that supports QoS (Quality of Service) for VoIP.

When I call someone or they call me, voice is only heard in one direction

1. The cause of this problem is usually a result of a router being misconfigured with respect to port mappings. Ensure all required VoIP ports are mapped to your TalkSwitch unit. Also, ensure you use a static private IP address when connected to your router as this can also affect port mappings from the firewall/router.
2. Use a router that supports uPnP.

For more VoIP information, see *VoIP Information* on page 231.

SUPPORT

If you are having problems with the configuration or operation of TalkSwitch, please read this section. If you require further assistance, do one of the following:

1. Contact your authorized TalkSwitch reseller.
2. Check the FAQ section located in the support area for your region at our website at **http://global.talkswitch.com**.
3. Contact our customer support department by e-mail: **support@talkswitch.com**.
4. Call a technical support specialist at TalkSwitch: **1.866.393.9960** (United States and Canada) or **613.725.2466** weekdays between 9:00 a.m. and 8:00 p.m. Eastern.

Please have the following information available when you call:

- Your e-mail address and phone number.
- The serial number located on the bottom of TalkSwitch.
- Your date of purchase.
- The number of telephone lines you have connected to TalkSwitch.
- The telephone company services you are using with TalkSwitch.
- Your PC's operating system (e.g. Windows Vista, XP, 2000).

TO OBTAIN WARRANTY SERVICE

For products purchased in the U.S. or Canada

1. Contact your reseller and obtain a WarrantyReturn Authorization (WRA) number if required. If you bought the product directly from TalkSwitch, you can obtain this number by contacting the Technical Support department at TalkSwitch — call toll free in the United States and Canada: (866) 393-9960, otherwise call (613) 725-2466. Please have your model and serial number available when you call.

   IMPORTANT: Failure to obtain a WRA numberand include the information indicated abovewill result in the package being refused.

2. Include a copy of your proof of purchase and a written explanation of the problem; this will enable us to expedite your repair.
3. Attach the shipping label to the package exterior.
4. On the shipping label, be sure to include the WRA number obtained from TalkSwitch or your reseller.
5. Ship the product (prepaid) to your reseller. Ifthe product was purchased from TalkSwitch, send the unit in its *original or better protective packaging* to:

   TalkSwitch
   1545 Carling Ave., Suite 510,
   Ottawa, Ontario K1Z 8P9
   Canada

   If the product is being shipped from outside of Canada, include the following statement on the package: **'Canadian Goods Being Returned to Canada'**.

For products purchased outside of the U.S. and Canada

To be eligible to obtain warranty service during the warranty period, the Customer must provide proof of the date of purchase, serial number, and obtain warranty return instructions from a service facility designated by TalkSwitch.

To obtain warranty service, the Customer is responsible for the cost of all shipping charges, duties and taxes for products sent to the authorized service facility. Replaced or repaired product is warranted to be free from defects in material or workmanship for the remainder of the limited warranty period of the originally purchased Product.

We trust your **TalkSwitch telephone system** will provide exceptional features, performance and value to your business. Should you have any further questions, please contact your authorized TalkSwitch reseller. If you purchased directly from TalkSwitch, visit the support section at **http://global.talkswitch.com**.

We welcome your feedback, comments and suggestions. Please e-mail us at **info@talkswitch.com** or write us at **TalkSwitch, 1545 Carling Avenue, Suite 510, Ottawa, ON Canada K1Z 8P9**.

Thank you for choosing TalkSwitch.

APPENDIX A

EXTENSION NUMBERS, FUNCTIONS AND COMMANDS

The following tables list extension numbers, functions and commands you can enter with a phone.

EXTENSION NUMBERS

The following table lists the local extensions, remote extensions, general voice mailboxes, VoIP numbers, and ring groups. Note that dialing 101–108 will ring local extensions 111–118.

	Unit 1	Unit 2	Unit 3	Unit 4
Local extensions (analog or IP)	111–118	121–128	131–138	141–148
IP extension Local extensions (IP only)	151–158	161–168	171–178	181–188
Remote extensions	210–219	220–229	230–239	240–249
General voice mailboxes	410–419	420–429	430–439	440–449
VoIP VoIP numbers	There are 50 VoIP numbers for a TalkSwitch VoIP network: 250–299.			
Ring groups	There are ten ring groups for the entire system: 300–309.			

FUNCTIONS YOU CAN ENTER FROM LOCAL EXTENSIONS

IP extension The following table lists functions that can be entered from local extensions. On an analog extension, press **Flash** or **Recall**. On an IP extension, press **Hold**.

Keys	Function performed
Flash or **Recall/Hold**	If you are connected to a caller, **Flash** or **Recall/Hold** puts the call on hold and you hear the internal dial tone.
Flash or **Recall/Hold**	Pressing **Flash** or **Recall/Hold** again at the internal dial tone retrieves the most recent call on hold.
Flash or **Recall/Hold** at external dial tone	When you are connected to an outside line, pressing **Flash** or **Recall/Hold** switches you to the internal dial tone.
* and mailbox number	Dials into a local, remote or general voice mailbox to leave a message.
** and mailbox number	Accesses a local, remote or general voice mailbox. You can retrieve messages, change greetings, password, etc.
** #	Accesses the mailbox associated with your extension.
Flash or **Recall/Hold 4**	Completes a transfer and returns to the internal dial tone.
IP extension On an analog extension, press **Flash 4** or **Recall 4**. On an IP extension, press **Hold 4** followed by **Dial, Send** or **#** to complete the transfer.	

0 (**9** in some regions)	Dials the receptionist.
***0**	External PA access to make an announcement through the attached external PA system.
***300–399**	Dials the system speed dial number.
Flash or **Recall/Hold 5**	Disconnects the current caller and reconnects to the last caller on hold. A conference-call initiator can disengage the second conferenced party by pressing **Flash 5** or **Recall 5**.
Flash or **Recall/Hold *500–509**	Press **Flash** or **Recall/Hold** to place a call on hold and then press ***500–509** to place the call in one of ten park orbits.
Flash or **Recall/Hold *510**	Press **Flash** or **Recall/Hold** to place a call on hold and then press ***510** to place the call in the next available park orbit, starting at 500.
****500–509**	Retrieves a parked call.
Flash or **Recall/Hold 6**	Conference call for 3 parties. Press **Flash** or **Recall** to put the first caller on hold, dial another extension or external number and press **Flash 6** or **Recall 6**.
***56**	TalkSwitch unit will announce its IP address.
***60**	Disables Do Not Disturb at the current extension.
***61**	Enables Do Not Disturb at the current extension.
***62**	Toggles Do Not Disturb at the current extension.
***7** and extension	Call pick-up at a specific extension. Dial from dial tone to pick up a call ringing at another extension.
Flash or **Recall/Hold 7**	Press **Flash 7** or **Recall 7** to cycle through queued calls on a first in/first out basis.
80	Accesses Same Line Connect when transferring an outside caller to another outside number.
9 (**0** in some regions), **81–88**	Dial to access an outside line.
On an IP extension, press **Dial, Send** or **#** after your entry for the following options:	
***80**	Play music on hold from an internal .wav file or an external audio source through the PA jack. If music on hold is playing through the PA jack, and a user dials ***0** to make an overhead page, the overhead page will interrupt the music on hold. However, if the PA jack is configured to perform voicemail screening, and a user or caller accesses a voice mailbox, music on hold will not be interrupted.
***810#**	Display the alternate name instead of the caller ID name.
***81 [1–10] #**	Route future calls from the last caller according to the specified group.
***81255#**	Delete the last caller from the caller ID lookup list.
***84** and extension	Hands-free intercom call to an extension, automatically answered in speaker mode, enabling instant hands-free 2-way communication (for supported sets).
***85** and extension or ring group number	Page a phone at an extension or the phones of a ring group.
***88** and account code	Attach account code to call detail record (CDR) for last call.

***9**	Call pick-up of any ringing line.
# on an analog extension phone, or ***55#** on a TalkSwitch IP phone (note: other brands may use ***55 Send** or ***55 Dial**).	Enters command mode.

FUNCTIONS YOU CAN ENTER FROM OUTSIDE PHONES

The following table lists functions that can be entered from outside phones after the auto attendant answers.

Keys	Function performed
0 (**9** in some regions), **1**, **2**, **3**, **4** and **5**	Performs the action defined in the auto attendant configuration.
3-digit local extension	Rings at the local extension (e.g. 111).
3-digit remote extension	Rings at the remote extension (e.g. 211).
3-digit VoIP number	Rings the VoIP number (e.g. 250). There are 50 VoIP numbers in a TalkSwitch VoIP network.
3-digit ring group	Rings all the extensions in the ring group (e.g. 301). There are ten ring groups for the entire system.
61	Causes the system to call you back at the current prompted call back number, after you hang up.
62	Causes the system to prompt you for your password and then the new prompted call back number. After you hang up, the system calls you back.
7 and extension	Transfers a caller on hold to another phone system used in conjunction with TalkSwitch. As soon as the call connects to the other system, it disconnects from TalkSwitch.
9 (**0** in some regions) and **81–88**	Accesses a hunt group for call bridge and is protected with the call bridge password.
80	Accesses call bridge using Same Line Connect. Requires 3–way calling/conference on the line.
* and mailbox number	Dials into a local, remote or general voice mailbox to leave a message.
** and mailbox number	Accesses voicemail of a local, remote or general voice mailbox. You can retrieve messages, change greetings, password, etc.

FUNCTIONS YOU CAN ENTER IN COMMAND MODE

The following table lists functions you can enter in command mode from a local extension or an outside phone. Enter command mode by either pressing **#** on an analog extension phone, or ***55#** on a TalkSwitch IP phone (note: other brands may use ***55 Send** or ***55 Dial**). When prompted, enter the system password + **#**. You can then enter the command.

Keys	Function performed
00 + **#**	TalkSwitch tell you the Unit ID of the unit you are connected to.
0 and ID number + **#**	Assigns a Unit ID (1 to 4).

30 + **#**	TalkSwitch tells you which mode it is currently using.
3 and mode number + **#**	Switch modes (1 = mode 1, 2 = mode 2, 3 = holiday mode).
4 and auto attendant number + **#**	Record auto attendant message (1 to 9).
04 and auto attendant number + **#**	Erase auto attendant message (1 to 9).
5 and auto attendant number + **#**	Play auto attendant message (1 to 9).
6 and auto call back number + **#**	Record auto call back announced message (1 to 4).
06 and auto call back number + **#**	Erase auto call back announced message (1 to 4).
7 and auto call back number + **#**	Play auto call back announced message (1 to 4).
65 + **#**	Record prompted call back announced message.
065 + **#**	Erase prompted call back announced message.
75 + **#**	Play prompted call back announced message.
90 + **#**	Play unit IP address.
Remote extension number, **8**, and phone number + **#**	Set a remote extension phone number.
Remote extension number, **980**, and **0** + **#**	Turn off Same Line Connect for a remote extension.
Remote extension number, **980**, and **1** + **#**	Turn on Same Line Connect for a remote extension.
Remote extension number, **9**, and hunt group + **#**	Set the hunt group for a remote extension.
Local extension number and **61** + **#**	Enable Do Not Disturb at the local extension.
Local extension number and **60** + **#**	Disable Do Not Disturb at the local extension.

APPENDIX B

TALKSWITCH AND TELEPHONE COMPANY SERVICES

Telephone companies offer a range of services on their telephone lines. The range of services varies by telephone company, and some services may not be available on all lines. Please consult your local telephone company for feature availability. Many of the services offered by your local telephone company are compatible with and enhanced by TalkSwitch's capabilities.

Use of services may require a subscription or incur service charges.

CALLER ID

The Caller ID service displays the phone number and name associated with an incoming call. Caller ID is also referred to as Caller Identity, Call Display, Caller Display, Calling Number Display, or IdentiCall.

The TalkSwitch unit can pass the caller ID to its local extensions when incoming calls are transferred by the auto attendant. If a local extension is configured to ring at least twice before the auto attendant answers, TalkSwitch will pass the caller ID to that extension. If a call is transferred from one extension to another, the caller ID will also be passed on. The TalkSwitch unit can use the time stamp from the caller ID to set its internal clock.

Analog extension telephone sets connected to TalkSwitch must support FSK type caller ID. TalkSwitch is compatible with telephone company FSK, DTMF and Dual Mode caller ID.

DISTINCTIVE RING

The Distinctive Ring service provides multiple phone numbers on one telephone line. Each phone number causes the phone to ring with a different ring pattern. The ring patterns have one, two, or three rings in each burst. The ring pattern identifies the number that was dialed. The service allows a different phone number for a fax machine or home-based business on one telephone line.

Distinctive ring is also referred to as Ident-A-Ring, Ident-A-Call, Ring Master, Teen Service, Double Number, Personal Ringing, Ring Plus, Smart Ring Service, Special Ring, or Customized Ringing. Call Sign™, My Ring, Duet® - Phone and Fax Multiple Number, or Smart Ring.

The TalkSwitch unit supports a main phone number and two distinctive ring phone numbers on each telephone line. It can detect the distinctive rings and use them for:

- Routing fax, modem and voice calls to a fax machine, modem or auto attendant.
- Separating home and office calls.
- Using auto call back.

TalkSwitch may not be compatible with some distinctive ring patterns.

VOICEMAIL

The Voicemail service gives callers the opportunity to leave a message when your telephone line is busy or unanswered. Voicemail is also referred to as Call Answer. TalkSwitch can work in conjunction with your telephone company's Voicemail service.

CALL WAITING

The Call Waiting service beeps when you have another call waiting to be answered.

To talk to the other caller when you hear a call waiting beep, press the **Flash** or **Recall** button twice. To return to the other caller, press the **Flash** or **Recall** button twice again.

The Call Waiting service is not recommended, since TalkSwitch cannot answer the second call while the first call is already being routed by the auto attendant. You may want to consider adding the Hunt/Rollover service to your lines instead.

3-WAY CALLING/CONFERENCE

The 3-Way Calling/Conference service allows you to add a third party to a call already in progress. It permits the Same Line Connect feature to be used with call forwarding, call back and call bridge (DISA), whether you have one or more telephone lines. With Same Line Connect, TalkSwitch can use the same telephone line to connect a second external caller.

If you are having difficulties with line volume levels when you forward calls to remote extensions or when you are using call bridge, forward calls using 3-Way Calling/Conference.

The 3-Way Calling/Conference service is not recommended with automatic route selection (ARS) or toll restriction. The TalkSwitch unit cannot route or block calls established through 3-Way Calling/Conference.

TalkSwitch is compatible with a variety of 3-Way Calling/Conference services. Where multiple services are available, you can designate the service type on a per line basis. TalkSwitch may not be compatible with some 3-Way Calling/Conference services.

TRANSFER AND CLEAR

The Transfer and Clear service allows you to add a third party to a call already in progress. Transfer and Clear is similar to 3-Way Calling/Conference, but allows the two external parties to remain in conversation after you hang up. Transfer and Clear is also known as FeatureLine or Centrex/Plexar.

If you have the Transfer and Clear service, you must enable the Transfer and Clear feature. Select the *Clear telephone line after call transferred* check box in the *To Remote Extensions* tab of the **Transfer Preferences** window. You must also select the *Transfer and clear* check box in the **Telephone Lines** page.

With Transfer and Clear, the TalkSwitch unit can release the telephone line after transferring a call to a remote extension. Otherwise the telephone line will remain occupied while the caller is connected to the remote extension.

If you don't have the service but enable the feature, the caller will be disconnected when the unit attempts to forward their call to the remote extension.

The Transfer and Clear service may require the remote extension to be answered before it can transfer the call and clear the line. If so, select one of the screening options in the call cascade of the remote extension. See *Remote Extensions* on page 115.

HUNT/ROLLOVER

The Hunt/Rollover service hunts across a set of telephone lines to find a free line for an incoming call. Two or more lines are required. This allows an organization with multiple telephone lines to publish one phone number. Hunt/Rollover is also known as Hunt Group, Busy Rollover, Hunting Facility, or Call Forward On Busy.

The TalkSwitch unit uses hunt groups for outgoing calls. If you know the order that the telephone company hunts across telephone lines for incoming calls, reverse the order in the hunt groups to prevent congestion.

OTHER SERVICES

Services using call back

Services which result in a call back to the TalkSwitch unit are not recommended, unless careful consideration is given to how the incoming call will be handled.

Services using prompted call back

If you call a number and it's busy, some telephone companies offer a prompted call back service that will notify you when the number is free. The use of this service is not recommended with TalkSwitch remote extensions.

If the service cannot be deactivated, set the *When a call is answered* list to *play accept/reject prompt* in the *Answered* tab of the remote extension's call cascade. This setting allows the call to be handled correctly if the telephone company answers the call to the remote extension with a prompted call back message.

Services and systems using Flash or Recall

Some services and systems require a **Flash** or **Recall** signal for activation. For example, a Key telephone system, PBX or telephone company service may require a **Flash** or **Recall** signal.

To generate a **Flash** or **Recall** signal for the service or system, press the **Flash** or **Recall** or **Link** button twice. Pressing the **Flash** or **Recall** button once will put the caller on hold, but the **Flash** or **Recall** signal will not be sent to the telephone line. This prevents the TalkSwitch system from hanging up on the caller. Pressing the **Flash** or **Recall** button a second time will send a **Flash** or **Recall** signal to the telephone line.

If your telephone does not have a **Flash** or **Recall** or **Link** button, you can generate a **Flash** or **Recall** signal by quickly pressing and releasing the telephone's hook switch.

If you need to send a **Flash** or **Recall** signal to the telephone line, press and release the hook switch once, count to two, and then press and release the hook switch a second time. If you initiate the second **Flash** or **Recall** signal too quickly, you will disconnect your caller.

Services using Ringback Tones

The Ringback Tones service allows the caller to hear a tune or announcement instead of the traditional ring back tone. This service is also known as Colour-Me Tones. The use of this service is not recommended with TalkSwitch remote extensions.

If the service cannot be deactivated, set the *When a call is answered* list to *play accept/reject prompt* in the *Answered* tab of the remote extension's call cascade. This setting allows the call to be handled correctly if the telephone company plays a tune or announcement instead of a traditional ring back tone.

Services using line reversals

In certain regions, telephone companies may use line reversal signaling to improve call handling on busy telephone lines. TalkSwitch units support various forms of line reversal, and allow per-line configuration of service type. TalkSwitch units are pre-set to work with the common line reversal service type for each region, where applicable.

APPENDIX C

TALKSWITCH AND POWER INTERRUPTIONS

TalkSwitch settings and configurations
In the event of a power failure, your TalkSwitch settings and configuration will not be lost.

The TalkSwitch configuration is stored in non-volatile memory. Non-volatile memory means your feature settings (configuration) remain in the TalkSwitch memory despite interruptions in the power supply.

Calling features
In the event of a power failure, each TalkSwitch unit will connect extension E4 to phone Line 1 to permit calls (except-AU models).

Calls must be dialed without a hunt group. All other TalkSwitch functions will not be available until power is restored. We recommend that a telephone that does not depend on mains power be available for emergency use.

Mode scheduling and power interruptions
When the power supply to TalkSwitch has been interrupted, TalkSwitch's internal clock will continue to run for up to 12 hours. If you have enabled the automatic mode scheduling feature and it has been longer than 12 hours, TalkSwitch will not function properly until the internal clock is reset via a caller ID call or with the TalkSwitch management software. When power is restored to TalkSwitch, it will check the time and determine the mode it should be running according to the configuration and set itself to that mode.

To check the current mode of operation:

1. Enter command mode by either pressing **#** on an analog extension phone, or ***55#** on a TalkSwitch IP phone (note: other brands may use ***55 Send** or ***55 Dial**).
2. Dial the system password + **#**.
3. Dial **30#**. TalkSwitch will report the current mode.

APPENDIX D

SAFETY AND REGULATORY INFORMATION

The following information applies to TalkSwitch units bearing the part number CT.TS001.1.

Safety Precautions
Before using TalkSwitch, please review and ensure the following safety instructions are adhered to:

1. Read, follow and retain instructions — All safety and operating instructions should be read, followed and retained for future reference before operating the equipment.
2. Heed warnings — All warnings on the equipment and in the operating instructions should be adhered to.
3. Temperature — Do not operate in environments where the temperature is below 0°C or 32°F. If the product was shipped in cold weather, please let the system warm up at room temperature for at least two hours before plugging it in to a power outlet.
4. Moisture — Do not place TalkSwitch in a high-humidity environment.
5. Heat — Never place TalkSwitch near heat sources such as radiators, floor registers or in direct sunlight. Do not stack units on top of each other.
6. Power supply — The equipment should only be connected to an approved power supply of the type described in the operating instructions or marked on the equipment. Use only the power adapter supplied with TalkSwitch.
7. Damage requiring service — Do not attempt to service the TalkSwitch yourself. Unplug TalkSwitch and refer servicing to a licensed technician when:
    - The plug or power cord has been damaged.
    - TalkSwitch has been exposed to moisture.
8. Emergency services — If you call an emergency service using an external IP extension, the TalkSwitch system will not send the address of your location. You must provide your address to the emergency operator.

External audio source isolation
We recommend that any external audio source be connected to the TalkSwitch unit using an approved Line Isolation Unit (e.g. A-Tick ( ) approved in AU).

Call Redirection & Service Provider Billing Advisory

Use of the automatic route selection, toll restriction and Call Detail Recording features does not imply any guarantee whatsoever by regulatory authorities, your telephone service provider(s), TalkSwitch or its distributors and resellers, with regard to the accuracy of these features and that the use of such a features may not be considered by a telephone company in any disputes which may arise regarding the accuracy of any subscriber's telephone account.

Important Notices

CE Compliance

The product models listed below comply with the essential requirements of the European "Radio and Telecom Terminal Equipment" (R&TTE) directive 1999/5/EC, and are for connection to regular telephone lines (PSTN).This equipment has been tested and found to comply with the following standards:

EN55022:1998 + amendment A1:2000 + amendment A2:2003, Class B
EN55024:1998 + amendment A1:2001 + amendment A2:2003
EN60950–1:2001, first edition
EN61000–3-2:2000
EN61000–3-3:1995+ amendment A1:2001
CISPR22:1997 + amendment A1:2000 + amendment A2:2002

This equipment is marked with the CE symbol, indicating compliance with CE standards.

For Declaration of Conformity please contact local distributor or visit **http://global.talkswitch.com** and select the required country.

FCC and IC Warnings

This equipment complies with ACTA TIA/EIA/IS-968A and Part 68 of the Federal Communications Commission (FCC) rules in the United States. It also complies with regulations RSS210 and CS-03 of Industry Canada and Science Canada. This equipment also complies with Part 15 of the FCC Rules, as well as ICES003 of Industry Canada. Operation is subject to the following two conditions: (1) this device may not cause harmful interference, and (2) this device must accept any interference received, including interference that may cause undesired operation of the device.

FCC Part 15

This equipment has been tested and found to comply with the limits for a "CLASS B" Digital Device pursuant to Part 15 of the FCC rules. These limits are designed to provide a reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy, and if not installed and used in accordance with these instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment to an outlet on a circuit different to that which the receiver is connected.
- Consult the dealer or an experienced Radio/TV technician for help.

This CLASS B Digital apparatus meets all requirements of the Canadian interference-causing equipment regulations.

Changes or modifications not expressly approved by TalkSwitch could void the user's authority to operate the equipment.

ACTA TIA/EIA/IS-968A, FCC Part 68 and IC CS-03
Notice: The Industry Canada label identifies certified equipment. This certification means the equipment meets telecommunications network protective, operational and safety requirements as prescribed in the appropriate Terminal Equipment Technical Requirements document(s). The department does not guarantee the equipment will operate to the user's satisfaction.

Before installing this equipment, users should ensure it is permissible to be connected to the facilities of the local telecommunications company. The equipment must also be installed using an acceptable method of connection. The customer should be aware that compliance with the above conditions may not prevent degradation of service in some situations.

Repairs to certified equipment should be coordinated by a representative designated by the supplier. Any repairs or alterations made by the user to this equipment, or equipment malfunctions, may give the telecommunications company cause to request the user to disconnect the equipment.

Users should ensure, for their own protection, that the electrical ground connections of the power utility, telephone lines and internal metallic water pipe system, if present, are connected together. This precaution may be particularly important in rural areas.

Caution: Users should not attempt to make such connections themselves, but should contact the appropriate electric inspection authority, or electrician, as appropriate.

A label is located on the underside of the base unit containing both the FCC registration number and Ringer Equivalency Number (REN) or the IC registration number and Load Number. You must, upon request, provide this information to your local telephone company.

The REN is used to determine the quantity of devices that may be connected to the telephone line. Excessive RENs on the telephone line may result in the devices not ringing in response to an incoming call. In most, but not all areas, the sum of RENs should not exceed five (5). To be certain of the number of devices that may be connected to a line, as determined by the total RENs, contact the local telephone company.

This equipment is compatible with inductively-coupled hearing aids.

If the equipment is causing harm to the telephone network, the telephone company may request that you disconnect the equipment until the problem is corrected.

This equipment cannot be used on public coin telephone services provided by the telephone company. Connection to party line service is subject to state tariffs.

The FCC requires that you connect your cordless telephone to the nation-wide telephone network through a modular telephone jack (USOC, RF11C, RJ11W, or RJ14).

Your telephone company may discontinue your service if your equipment causes harm to the telephone network. They will notify you in advance of disconnection, if possible. During notification, you will be informed of your right to file a complaint to the FCC.

Occasionally, your telephone company may make changes in its facilities, equipment, operation, or procedures that could affect the operation of your equipment. If so, you will be given advance notice of the change to give you an opportunity to maintain uninterrupted service.

ICASA
For all ICASA license enquiries please contact your South African authorized distributor for TalkSwitch.

Other Countries
For latest information on regulatory compliance in other markets see **http://global.talkswitch.com** and select required country.

RoHS Compliance
This equipment conforms with the Reduction of Hazardous Substances (RoHS) Directive 2002/95/EC. The RoHS directive provides for restriction or elimination of the following substances:

> Lead, mercury, cadmium, hexavalent chromium, polybrominated biphenyls (PBB), and polybrominated diphenyl ether (PBDE).

RoHS and related legislations are initiatives to reduce the environmental issues related to the manufacture of electronic equipment.

Disposal

At the end of the product's life, please ensure disposal is in compliance with local regulations for electrical and electronic waste.

APPENDIX E

TALKSWITCH ONE-YEAR WARRANTY AND RETURN POLICY

TalkSwitch limited warranty

Condition: The original end-user customer acknowledges that he or she has read the limited One-Year Warranty and Warranty Disclaimers and understands the same and agrees to be bound by its terms and conditions.

TalkSwitch, a division of Centrepoint Technologies Inc. ("TalkSwitch") warrants to the original end-user customer ("Customer") that each new TalkSwitch product shall be substantially free from defects in materials and workmanship under normal use for a period of one (1) year from the date of the original purchase (proof of purchase required). The exclusive remedy and entire liability under this warranty will be for repair or replacement on a like-for-like basis at TalkSwitch's option.

The above warranty shall not apply to product defects resulting from (a) improper maintenance or installation; (b) misuse, neglect or accident; (c) damage from moisture or corrosive environments; (d) use of the power adapter with supply voltages other than that for which it is specified; (e) high voltage surges; (f) electrostatic discharges; (g) operation outside the product's specification; or (h) failure to follow product instructions.

The warranty is void where (a) the serial number has been altered, removed or effaced; or (b) the product has been used in an application, country, region, locality, or connected to any network, other than those for which the product was intended to operate.

TalkSwitch shall not have any obligation to repair or replace product until the Customer returns defective product to TalkSwitch. Any replacement product may be either new or like-new, and may contain remanufactured parts, equivalent to new in performance.

EXCEPT FOR THE EXPRESS WARRANTIES SET FORTH ABOVE, AND TO THE EXTENT PERMITTED BY APPLICABLE LAW, THE PRODUCTS AND STORAGE MEDIA ARE SUPPLIED ON AN "AS IS" BASIS AND THERE ARE NO WARRANTIES, REPRESENTATIONS OR CONDITIONS WHETHER EXPRESSED OR IMPLIED, WRITTEN OR ORAL, ARISING BY STATUTE, OPERATION OF LAW, COURSE OF DEALING, USAGE OF TRADE OR OTHERWISE, WITH RESPECT TO THE PRODUCT. TALKSWITCH SPECIFICALLY DISCLAIMS ANY IMPLIED WARRANTIES OR CONDITIONS OF MERCHANTABILITY, SATISFACTORY QUALITY, DURABILITY, FITNESS FOR A PARTICULAR PURPOSE, AND NON-INFRINGEMENT.

IN NO EVENT SHALL TALKSWITCH BE LIABLE FOR ANY SPECIAL, INCIDENTAL, CONSEQUENTIAL, INDIRECT OR SIMILAR DAMAGES, INCLUDING ANY LOST PROFITS, LOST DATA, OR OTHER INCIDENTAL OR CONSEQUENTIAL OR PUNITIVE DAMAGES, WHETHER BASED ON CONTRACT, TORT (INCLUDING WITHOUT LIMITATION, NEGLIGENCE), OR ANY OTHER LEGAL THEORY, WHETHER OR NOT ADVISED OF THE POSSIBILITY OF SUCH DAMAGES, OR WHETHER SUCH DAMAGES ARE FORSEEABLE. IN ANY EVENT, THE TOTAL AGGREGATE LIABILITY OF TALKSWITCH FOR DIRECT DAMAGES SHALL NOT EXCEED THE PURCHASE PRICE OF THE PRODUCT.

This warranty gives specific rights, and Customer may have other rights subject to jurisdiction. To the extent any part of this limited warranty statement is inconsistent with such local law, that part shall be deemed modified to be consistent with such local law.

DOC Compliance
This digital apparatus does not exceed the Class B limits for radio noise emissions for digital apparatus as set out in the Radio Interference Regulations of the Canadian Department of Communications.

The FCC Statement is located in *Appendix D: Safety and Regulatory Information*.

Warranty Service
For products purchased in the U.S. or Canada

1. Contact your reseller and obtain a Warranty Return Authorization (WRA) number if required. If you bought the product directly from TalkSwitch, you can obtain this number by contacting the Technical Support department at TalkSwitch — call toll-free in the United States and Canada: (866) 393-9960, otherwise call (613) 725-2466. Please have your model and serial number available when you call.

   IMPORTANT: Failure to obtain a WRA numberand include the information indicated abovewill result in the package being refused.

2. Include a copy of your proof of purchase and a written explanation of the problem; this will enable us to expedite your repair.

3. Attach the shipping label to the package exterior.

4. On the shipping label, be sure to include the WRA number obtained from TalkSwitch or your reseller.

5. Ship the product (prepaid) to your reseller. If the product was purchased from TalkSwitch, send the unit in its *original or better protective packaging* to:

   TalkSwitch
   1545 Carling Ave., Suite 510,
   Ottawa, Ontario K1Z 8P9
   Canada

   If the product is being shipped from outside of Canada, the following statement must be included on the package: **Canadian Goods Being Returned to Canada**.

For products purchased outside of the U.S. and Canada
To be eligible to obtain warranty service during the warranty period, the Customer must provide proof of the date of purchase, serial number, and obtain warranty return instructions from a service facility designated by TalkSwitch.

To obtain warranty service, the Customer is responsible for the cost of all shipping charges, duties and taxes for products sent to the authorized service facility. Replaced or repaired product is warranted to be free from defects in material or workmanship for the remainder of the limited warranty period of the originally purchased Product.

APPENDIX F

SPECIFICATIONS

The following specifications apply to TalkSwitch units bearing the part number CT.TS001.1.

Interfaces

	240vs	244vs	248vs	280vs	284vs	288vs
Telephone lines (FXO)	2	2	2	2	2	2
Analog extensions (FXS)	4	4	4	8	8	8
VoIP ports	0	4	8	0	4	8
VoIP numbers	0	12	24	0	12	24
Local extensions (Analog and IP)	16	16	16	16	16	16
	480vs	**484vs**	**488vs**	**840vs**	**844vs**	**848vs**
Telephone lines (FXO)	4	4	4	8	8	8
Analog extensions (FXS)	8	8	8	4	4	4
VoIP ports	0	4	8	0	4	8
VoIP numbers	0	12	24	0	12	24
Local extensions (Analog and IP)	16	16	16	16	16	16

Interface Specifications

LAN interface	RJ-45 Ethernet (10BaseT)
USB interface	Standard USB Type B
Memory	2 hours internal 1–, 2–, 4.5– and 9–hour memory cards extend internal memory, but may not be available in all markets.
Music input	Phono jack (1/8" mono) or stored .wav file
PA output	Phono jack (1/8" mono)
Power source	Input: 120 Vac 60 Hz or 220-240 Vac 50/60 Hz, depending on region Output: 16 Vac 60 Hz 2.0 A
Power-failure line to extension jacks	Extension 4 connects to Line 1. Note: Does NOT apply in AU.

Analog Extension Specifications

Connector type	RJ-11, 2-pin
Trunk type	Loop start
Interface impedance	Either 600 Ω, 900 Ω, TBR21, BT complex, AU complex or ZA complex, depending on region.
Loop range	0–600 Ω
Recall or Flash supported	Yes
On-hook voltage	35 V
Off-hook loop voltage source	35 V
Off-hook loop current	23 mA to 40 mA
Dial tone level	-2.4 dBm
Ringing voltage	90 V RMS
Ringing frequency	20 Hz or 25 Hz (depending on region) – True sine wave
Total ringing load	5 REN

Telephone Line Specifications

Connector type	RJ-11, 2-pin
Trunk type	Loop start
Interface impedance	Configurable, 600 Ω, 600 complex, 900 Ω, 900 complex
Ringing impedance	3,900 Ω
Ringing sensitivity	45 V RMS @ 20 Hz or 25 Hz (depending on region)
REN	0.7 B
Longitudinal balance	69 dB
Signal level	-10 dB
Distinctive ring supported	Yes
Caller ID support	FSK, all lines

Dimensions

Metric (mm)	51.6 x 203.2 x 213.4
Imperial (inches)	2.03 x 8.0 x 8.4

Weight

	240vs	**244vs**	**248vs**	**280vs**	**284vs**	**288vs**
Metric (kg)	0.65	0.67	0.67	0.72	0.74	0.74
Imperial (lbs)	1.43	1.46	1.46	1.58	1.62	1.62
	480vs	**484vs**	**488vs**	**840vs**	**844vs**	**848vs**
Metric (kg)	0.79	0.81	0.81	0.89	0.91	0.91
Imperial (lbs)	1.74	1.77	1.77	1.96	1.99	1.99

Regulatory Approvals

ACTA TIA/EIA/IS-968A and FCC Part 68, Industry Canada CS-03

FCC Part 15B, Industry Canada ICES-003

Power adapter complies with UL and CSA

Reduction of Hazardous Substances (RoHS) Directive 2002/95/EC

VoIP Features (models with VoIP lines only)

- Embedded SIP server and user agents
- Integrated registrar
- Fax tone detection
- RTP
- G.711µ-law/A-law, G.726 (32 Kbps), G.729a
- G.168 echo cancellation
- VAD, silence suppression
- Adaptive jitter buffer
- Packet loss concealment
- Can use external SIP servers as required

APPENDIX G

COPYRIGHT AND LICENSING NOTICES

© 2009 TalkSwitch, a division of Centrepoint Technologies Inc. All Rights reserved.
TalkSwitch software is protected under copyright law and international treaties. Unauthorized reproduction or distribution of TalkSwitch software, or any portion of it, in any form, may result in severe civil and criminal penalties, and will be prosecuted to the maximum extent of the law.

TalkSwitch®, the TalkSwitch logo, Concero®, answers with intelligence®, owner friendly™, seller friendly™, vendor friendly™ and *channel friendly™* are registered trademarks or trademarks of Centrepoint Technologies Inc. All other trademarks are the property of their respective owners.

Elements of TalkSwitch software are protected under additional copyrights listed below, and subject to the terms and disclaimers listed below.

A) © 1998 by the Massachusetts Institute of Technology.
Permission to use, copy, modify, and distribute this software and its documentation for any purpose and without fee is hereby granted, provided that the above copyright notice appear in all copies and that both that copyright notice and this permission notice appear in supporting documentation, and that the name of MIT not be used in advertising or publicity pertaining to distribution of the software without specific, written prior permission. MIT makes no representations about the suitability of this software for any purpose. It is provided "as is" without express or implied warranty.

B) © 2000–2003 Intel Corporation. All rights reserved.
Redistribution and use in source and binary forms, with or without modification, are permitted provided that the following conditions are met:

- Redistributions of source code must retain the above copyright notice, this list of conditions and the following disclaimer.
- Redistributions in binary form must reproduce the above copyright notice, this list of conditions and the following disclaimer in the documentation and/or other materials provided with the distribution.
- Neither name of Intel Corporation nor the names of its contributors may be used to endorse or promote products derived from this software without specific prior written permission.

THIS SOFTWARE IS PROVIDED BY THE COPYRIGHT HOLDERS AND CONTRIBUTORS "AS IS" AND ANY EXPRESS OR IMPLIED WARRANTIES, INCLUDING, BUT NOT LIMITED TO, THE IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE ARE DISCLAIMED. IN NO EVENT SHALL INTEL OR CONTRIBUTORS BE LIABLE FOR ANY DIRECT, INDIRECT, INCIDENTAL, SPECIAL, EXEMPLARY, OR CONSEQUENTIAL DAMAGES (INCLUDING, BUT NOT LIMITED TO, PROCUREMENT OF SUBSTITUTE GOODS OR SERVICES; LOSS OF USE, DATA, OR PROFITS; OR BUSINESS INTERRUPTION) HOWEVER CAUSED AND ON ANY THEORY OF LIABILITY, WHETHER IN CONTRACT, STRICT LIABILITY, OR TORT (INCLUDING NEGLIGENCE OR OTHERWISE) ARISING IN ANY WAY OUT OF THE USE OF THIS SOFTWARE, EVEN IF ADVISED OF THE POSSIBILITY OF SUCH DAMAGE.

C) © 2001–2003 Swedish Institute of Computer Science. All rights reserved.
Redistribution and use in source and binary forms, with or without modification, are permitted provided that the following conditions are met:

- Redistributions of source code must retain the above copyright notice, this list of conditions and the following disclaimer.
- Redistributions in binary form must reproduce the above copyright notice, this list of conditions and the following disclaimer in the documentation and/or other materials provided with the distribution.
- The name of the author may not be used to endorse or promote products derived from this software without specific prior written permission.

THIS SOFTWARE IS PROVIDED BY THE AUTHOR "AS IS" AND ANY EXPRESS OR IMPLIED WARRANTIES, INCLUDING, BUT NOT LIMITED TO, THE IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE ARE DISCLAIMED. IN NO EVENT SHALL THE AUTHOR BE LIABLE FOR ANY DIRECT, INDIRECT, INCIDENTAL, SPECIAL, EXEMPLARY, OR CONSEQUENTIAL DAMAGES (INCLUDING, BUT NOT LIMITED TO, PROCUREMENT OF SUBSTITUTE GOODS OR SERVICES; LOSS OF USE, DATA, OR PROFITS; OR BUSINESS INTERRUPTION) HOWEVER CAUSED AND ON ANY THEORY OF LIABILITY, WHETHER IN CONTRACT, STRICT LIABILITY, OR TORT (INCLUDING NEGLIGENCE OR OTHERWISE) ARISING IN ANY WAY OUT OF THE USE OF THIS SOFTWARE, EVEN IF ADVISED OF THE POSSIBILITY OF SUCH DAMAGE.

D) © 1991–1992, RSA Data Security, Inc. Created 1991. All rights reserved.
License to copy and use this software is granted provided that it is identified as the "RSA Data Security, Inc. MD5 Message-Digest Algorithm" in all material mentioning or referencing this software or this function. License is also granted to make and use derivative works provided that such works are identified as "derived from the RSA Data Security, Inc. MD5 Message-Digest Algorithm" in all material mentioning or referencing the derived work. RSA Data Security, Inc. makes no representations concerning either the merchantability of this software or the suitability of this software for any particular purpose. It is provided "as is" without express or implied warranty of any kind. These notices must be retained in any copies of any part of this documentation and/or software.

E) © 2004 Magnus Damm <damm@opensource.se> All rights reserved.
Permission is hereby granted, free of charge, to any person obtaining a copy of this software and associated documentation files (the 'Software'), to deal in the software without restriction, including without limitation the rights to use, copy, modify, merge, publish, distribute, sublicense, and/or sell copies of the software, and to permit persons to whom the software is furnished to do so, subject to the following conditions:

The above copyright notice and this permission notice shall be included in all copies or substantial portions of the software.

THE SOFTWARE IS PROVIDED "AS IS", WITHOUT WARRANTY OF ANY KIND, EXPRESS OR IMPLIED, INCLUDING BUT NOT LIMITED TO THE WARRANTIES OF MERCHANTABILITY, FITNESS FOR A PARTICULAR PURPOSE AND NONINFRINGEMENT. IN NO EVENT SHALL THE AUTHORS OR COPYRIGHT HOLDERS BE LIABLE FOR ANY CLAIM, DAMAGES OR OTHER LIABILITY, WHETHER IN AN ACTION OF CONTRACT, TORT OR OTHERWISE, ARISING FROM, OUT OF OR IN CONNECTION WITH THE SOFTWARE OR THE USE OR OTHER DEALINGS IN THE SOFTWARE.

F) © 1991, 1999 Free Software Foundation, Inc.
3rd-Party IP Phone configuration library.
© 2006 Centrepoint Technologies, Inc.
This library is free software. You can redistribute it and/or modify it under the terms of the GNU Lesser General Public License as published by the Free Software Foundation, either version 2.1 of the License, or (at your option) any later version. This library is distributed in the hope that it will be useful, but WITHOUT ANY WARRANTY; without even the implied warranty of MERCHANTABILITY or FITNESS FOR A PARTICULAR PURPOSE. See the GNU Lesser General Public License for more details. You should have received a copy of the GNU Lesser General Public License along with this library. If not, write to the Free Software Foundation, Inc., 59 Temple Place, Suite 330, Boston, MA 02111–1307 USA.

Some navigation icons are provided by famfamfam.com and used under the Creative Commons Attribution 2.5 License as specified by the owner.

APPENDIX H

HOME/OFFICE WIRING GUIDE

This appendix describes phone wiring for North America, and is provided as reference information for other countries and regions.

For users in the United Kingdom, see page 11 for wiring information.

Basic residential phone wiring

Looped wiring

Loop wiring is common in most residential houses. The demarcation point (see next page) breaks the incoming phone lines into 'loops' that can run the length of the entire house, but are often broken into smaller loops that serve different areas of the house.

All houses have at least one telephone line, however it is also common to have two or more incoming phone lines. Whether you are wiring single, dual or multiple lines, you will still be using phone cord with four wires. How each phone jack is wired to the loops will determine if the phone jack is on Line 1 or Line 2.

This document describes single- and two-line wiring configurations. Many of the processes can also be applied to multi-line wiring.

Single-line wiring

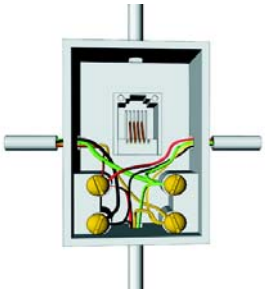
For a single telephone line, all phones in the building will usually be wired using the RED and GREEN wires of the phone cord. The other two wires, BLACK and YELLOW, are not used.

The demarcation point connects the incoming RED and GREEN pair to various loops within the building. It is probable that the building is also wired with a BLACK and YELLOW pair, however these are not connected to any wall jacks.

Two-line wiring

Two incoming phone lines are denoted in pairs, usually the RED and GREEN are used as Line 1, and the BLACK and YELLOW are Line 2. All four wires are looped throughout the building so that each jack has access to Line 1 and Line 2. How each jack is connected to the RED, GREEN, BLACK, and YELLOW wires determines if it is connected to Line 1, Line 2, or both.

Demarcation point

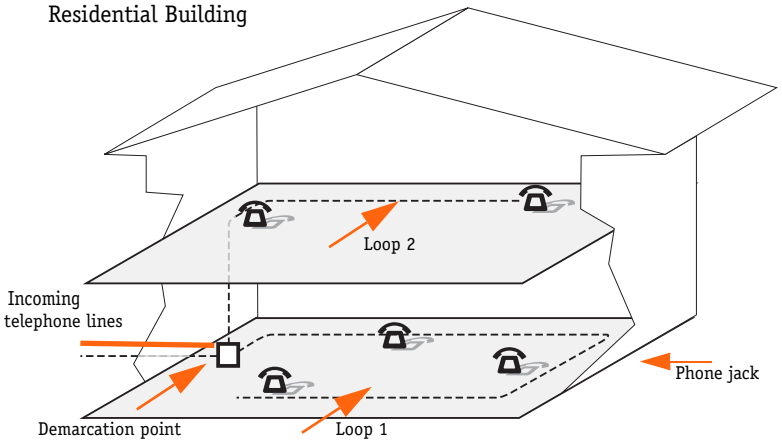


The demarcation point is the point of interconnection between the telephone company facilities and your building.

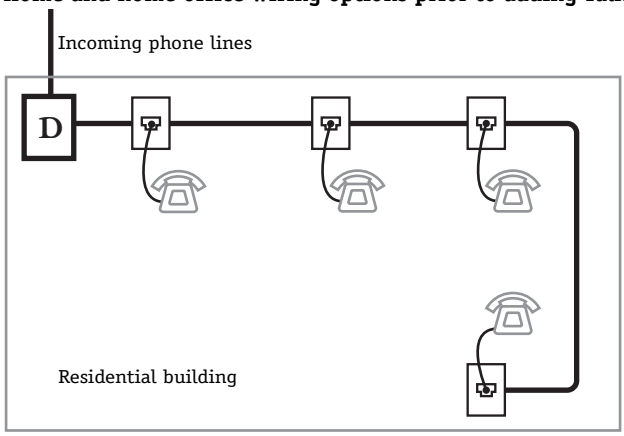
Location: It is usually a box located inside the building, within 15 feet of the incoming phone wiring. In newer houses, it may be located near the fuse box.

Wiring: The box will contain at least four posts used to branch the incoming wiring to the internal phone wiring. The RED post should have all the RED wires attached to it, the GREEN should have all the GREEN wires attached etc.

This demarcation point shows two loops. Each loop has two lines (four wires). Depending on how the jack is wired, the phone can be plugged into Line 1, Line 2, or both.



Home and home office wiring options prior to adding TalkSwitch



All phones are connected together in a single loop (or in multiple loops)

The phones are all wired to Line 1. If you have two or more incoming lines, the phones can be wired to Line 1, Line 2 or even both lines for two-line phones.

Prior to adding a TalkSwitch, all phones in the home ring in unison when a call comes in. Home phones do not have TalkSwitch functionality and rely on phone company features for any enhancements.

Choose a wiring method (Option 1 or Option 2 below) that best suits your needs.

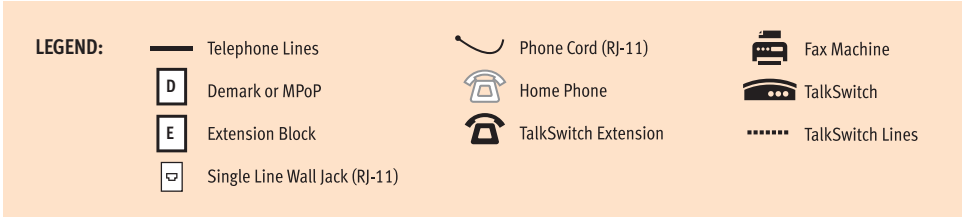
OPTION 1 — TalkSwitch phones separate from home phones

Advantages

- Easy to set up. May not require any wiring changes when used with one incoming line.
- You can still use some of TalkSwitch's features from the home phones.

Disadvantages

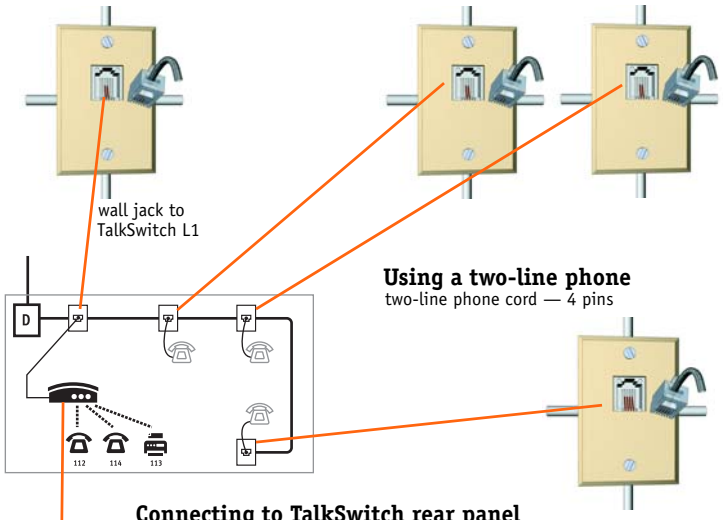
- Your home phones and office phones are not fully integrated together. For example, calls to the office cannot be forwarded to the home phones.
- Home phones don't have access to voicemail and other TalkSwitch features.
- All phones ring at the same time for incoming calls.



Connecting to phone lines

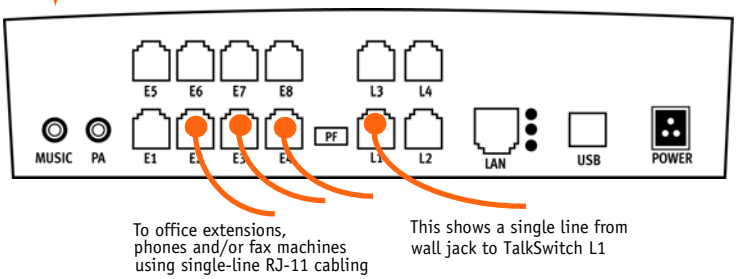
Using one telephone line
single-line phone cord — 2 pins

Using single-line phones
single-line phone cords — 2 pins



Using a two-line phone
two-line phone cord — 4 pins

Connecting to TalkSwitch rear panel
If you have two lines, you will need to connect them to jack L1 and jack L2.



To office extensions, phones and/or fax machines using single-line RJ-11 cabling

This shows a single line from wall jack to TalkSwitch L1

LEGEND:

- Telephone Lines
- D — Demark or MPoP
- E — Extension Block
- Single Line Wall Jack (RJ-11)
- Phone Cord (RJ-11)
- Home Phone
- TalkSwitch Extension
- Fax Machine
- TalkSwitch
- TalkSwitch Lines

OPTION 2 — TalkSwitch controls incoming lines and phones

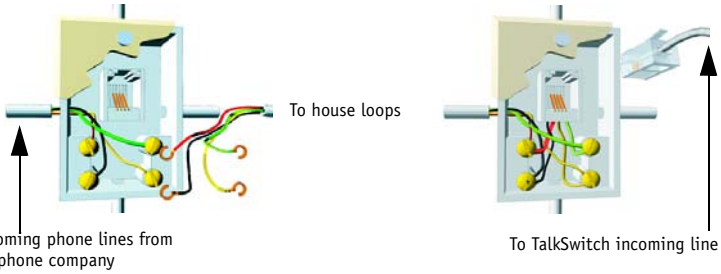
Advantages

- Home phone system and office phone system are integrated together.
- Calls to the home can be forwarded to the office and vice versa.
- The home phones have all the call handling capability as the office phones including voicemail, call cascade, mode scheduling, call forwarding etc.
- Suitable for one or two incoming phones lines.

Disadvantages

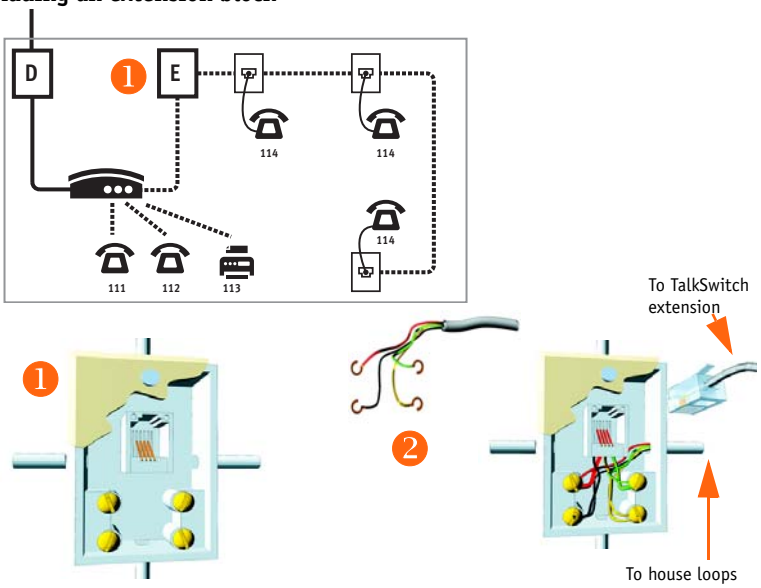
- Requires a little more wiring, and additional cabling.

Re-wiring the demarcation point



1. Loosen the screws connecting the wires to the posts.
2. Disconnect all house loops.
3. Connect a phone cable that will run to the extension block.
4. Tighten the screws connecting the wires to the posts.

Adding an extension block

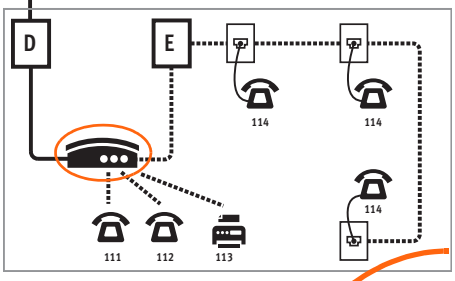


1. Add an extension block next to the demarcation point.
2. Connect the house loops and the phone cable from the demarcation point to the posts of the extension block.

LEGEND:

- Telephone Lines
- D — Demark or MPoP
- E — Extension Block
- Single Line Wall Jack (RJ-11)
- Phone Cord (RJ-11)
- Home Phone
- TalkSwitch Extension
- Fax Machine
- TalkSwitch
- TalkSwitch Lines

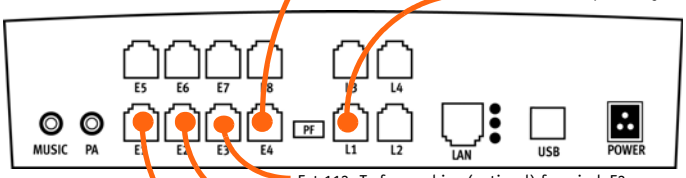
Connecting to TalkSwitch



To extension block, use a single-line phone cable from jack E4

Connecting to TalkSwitch rear panel

From demarcation point, use a single-line phone cable from the demarcation point to jack L1



Ext 113: To fax machine (optional) from jack E3

Ext 112: To phone from jack E2

To office extensions, phones and fax machines

Ext 111: To phone from jack E1

RJ-11 cabling can go up to 100 feet from the TalkSwitch without affecting voice quality. You may extend the lines farther although you may suffer voice quality.

Cabling can be run along baseboards, moldings, threaded between walls, and hidden under carpets. Keep the cords in corners and along edges to help hide it, and avoid high traffic areas.

1. Use a single-line RJ-11 phone cord to connect the demarcation point to TalkSwitch L1.
2. Use a single-line RJ-11 phone cord to connect the extension block to TalkSwitch E4.
3. Use single-line RJ-11 phone cords to connect the office phones to TalkSwitch E1, E2 and E3.

LEGEND:

- Telephone Lines
- D — Demark or MPoP
- E — Extension Block
- Single Line Wall Jack (RJ-11)
- Phone Cord (RJ-11)
- Home Phone
- TalkSwitch Extension
- Fax Machine
- TalkSwitch
- TalkSwitch Lines

GLOSSARY

*** Star/asterisk sign**

The character at the bottom left of your touchtone telephone keypad.

 # Number/pound/hash sign

The character at the bottom right of your touchtone telephone keypad.

3-way calling/conference call

Connecting three parties together for one telephone conversation. TalkSwitch offers its own 3-way conference feature the same as the 3-way conference service offered by telephone companies.

.wav format

This file format is 8000 Hz, 8-bit, mono, µ-law. This is the only file format that can be imported into TalkSwitch. Higher-quality formats, such as MP3, are not necessary since phone lines do not allow higher bandwidth and only support mono audio.

Analog

Analog devices transmit and receive sound in waves. Standard phones, cordless phones, answering machines, fax machines, modems, and the telephone lines in most homes are analog.

Call back

If call back is activated, it instructs TalkSwitch to call you back at a preset out-of-office phone number. With call back you can access the TalkSwitch long-distance plan while you are away. TalkSwitch offers automatic call back and prompted call back.

Call bridge

Allows you to make long-distance calls from anywhere in the world over your PBX using an access number and an authorization code. Also known as DISA in some regions.

Call forward

Call forward is used to forward calls to one of the remote extensions.

Caller ID

A service provided by telephone companies, transmitting information about calls (caller's name, phone number, time of call, etc.).

Call waiting

Call waiting allows you to accept a second incoming call on a telephone line that is already in use. Call waiting allows you to toggle from one caller to the other. The TalkSwitch system has call waiting built-in. The Call Waiting service from the telephone company is not recommended, since the TalkSwitch system cannot answer the second call while the first call is being routed by the auto attendant.

Centrex/Plexar

Central office exchange service. The telephone company offers PBX features on a 'pay-per-service' basis. This service may not be available from all telephone companies. Centrex/Plexar call switching happens at a local telephone company instead of your company's premises.

CNG

A tone emitted by most fax machines to initiate contact with another fax machine.

CO

Central office of a telephone company.

Communications software

Software to send and receive data over telephone lines via modems.

Compression

Reduces the data size. It saves data storage space, bandwidth and transmission time.

Conference call

Connecting three or more parties for one phone conversation. They can be internal or external TalkSwitch parties or remote callers.

Configuration

The way the system is set up. The management software defines the settings of the TalkSwitch system. These settings determine what each TalkSwitch feature will do and when and how it will do it.

Connector

The end of a cable that plugs into a port or interface to connect one device to another.

Digit

A whole number from 0 to 9. The numbers on your telephone keypad are digits.

Digital

Digital devices convert analog waves to digital signals for storage and transmission. The digital signal is converted back to analog waves that you can hear, and for transmission over analog lines.

DISA

Direct Inward System Access. Allows you to make long-distance calls from anywhere in the world over your PBX using an access number and an authorization code. Also known as call bridge in some regions.

Distinctive ring

A service provided by some telephone companies to distinguish two different phone numbers using the same telephone line (i.e. fax machine). TalkSwitch can detect one, two and three ring bursts per ringing cycle to provide different handling for incoming calls.

Driver

The software that controls a device (printer, disk drive, modem, etc.). The driver acts as a translator between the device and the programs that use it.

DTMF

Dual-tone Multi-frequency. It is the system used by touchtone telephones. DTMF assigns two specific frequencies (tones) to each key so that it can be identified by a microprocessor.

Dual-line telephone

A telephone allowing you to make and receive calls on two separate telephone lines.

Extension

See *Local extension* or *Remote extension*.

FCC

Federal Communications Commission. The FCC is the federal regulatory agency for the United States, responsible for the regulation of interstate and international communications by radio, television, wire, satellite and cable.

Firmware

Firmware is a combination of software and hardware. Microchips that have data or programs recorded on them are called firmware.

Flash or **Recall button**

A button on your telephone. Pressing the **Flash** or **Recall** button has the same effect as pressing the **Link** button or pressing and releasing the hang-up hook on a telephone. All of these actions send a signal to TalkSwitch to put calls on/off hold.

Hunt

Hunt (Rollover) is a phone company feature that allows calls to be forwarded to another line if the first line is busy. Most businesses with two or more lines subscribe to the rollover service so that only one phone number needs to be published for customers. You should configure all TalkSwitch lines that are part of the rollover group to answer in the same fashion.

Icon

A small picture representing a program, an action or a feature in the TalkSwitch management software. When you click on an icon, the program, action or feature is activated.

IETF

Internet Engineering Task Force. Develops and promotes Internet standards like TCP/IP and SIP protocols.

Intercom

A dedicated voice service within a specified user environment. TalkSwitch offers intercom capabilities between its local extensions.

Interface

Connects two separate entities. A user interface is the part of a program that you see on your computer screen and that connects you to the computer.

IP address

Internet Protocol. The standard protocol used by devices to communicate over a computer network. The IP address is a unique number or name that identifies the device.

IP extension

An extension that has its own IP address utilizing the Internet protocol in connection with VoIP.

IP phone

See *IP extension*.

ISDN

Integrated Services Digital Network.

ISP

Internet Service Provider. Facilitates access to the Internet for end-users. Most ISPs are telephone companies. Their services include domain name registration, web hosting, leased line access, etc.

Keypad

The push buttons on a telephone.

Keypad commands

Combinations of digits used to configure TalkSwitch from a touchtone telephone.

LED

Light Emitting Diode. A device that lights up when electricity is passed through. TalkSwitch has five LEDs on its front panel letting you know which lines are in use and three on the back panel beside the LAN port.

Line 1, 2, 3, or 4

A jack or plug-in on the back panel of TalkSwitch which can accept an RJ-11 plug from a telephone line or PBX system. You can plug in a single-line telephone cable into this jack and it becomes a usable TalkSwitch line.

Link button

The **Link** button has the same effect as the **Flash** or **Recall** button or pressing and releasing the hang-up hook on most telephones. On some telephones, the function of the **Link** button is internal to the telephone and pressing it has no effect on TalkSwitch.

Local extension

A local extension is an analog extension plugged into one of the extension jacks on the back panel of your TalkSwitch, or an IP extension connected to the TalkSwitch by LAN.

Modem

Acronym for MODulator-DEModulator. A modem is a device that converts digital information from a computer into analog signals for transmission over telephone lines and vice-versa.

Modes

TalkSwitch supports different time and day modes of operation, providing different call management depending on the current mode of operation.

Monitor (TalkSwitch monitor)

A feature included with the TalkSwitch management software. The monitor provides a graphical representation of telephone traffic as it flows through the system.

Network

A group of two or more computer systems linked together.

PBX

Private Branch eXchange. A PBX is a private telephone network used within an enterprise. The PBX controls access to telephone lines and services. It allows many users to share a smaller number of telephone lines using extensions and have access to additional call handling benefits (call hold, intercom calling, etc.). A TalkSwitch unit is a small PBX.

Phone load

See *Ringing equivalency*.

POP3

Version 3 of the post office protocol (POP) is a standard Internet protocol for retrieving e-mail from a remote server. Most subscribers to an Internet service provider have e-mail accounts using POP3.

Prompt

Recorded instructions played to a caller by TalkSwitch. Prompts are also used to provide instructions when you configure TalkSwitch using a telephone keypad.

PSTN

Public Switched Telephone Network. Your telephone line is on a PSTN.

Rollover

Rollover (Hunt) is a phone company feature that allows calls to be forwarded to another line if the first line is busy. Most businesses with two or more lines subscribe to the rollover service so that only one phone number needs to be published for customers. You should configure all TalkSwitch lines that are part of the rollover group to answer in the same fashion

Registered jacks

Telephone and data plugs registered with the FCC.

Remote extension

A call forwarding number that can be a phone number in any location, set up as a virtual extension of the TalkSwitch unit. The remote extension is assigned an extension number and has a voice mailbox.

REN

Ringer Equivalency Number. A number determined in accordance with the US code of federal regulations. It represents the ringer loading effect on a line.

RJ-11

Registered jack 11 connects to a 2-wire telephone line. It is used with telephones, cordless phones, fax machines and modems.

RJ-14

Registered jack 14 connects to a 4-wire telephone line. It is used with two-line telephones.

RJ-45

Registered jack 45 connects to an 8-wire telephone line. It is generally used with networking applications.

Rollover

Rollover (Hunt) is a phone company feature that allows calls to be forwarded to another line if the first line is busy. Most businesses with two or more lines subscribe to the rollover service so that only one phone number needs to be published for customers. You should configure all TalkSwitch lines that are part of the rollover group to answer in the same fashion.

RTP

Real-time Transport Protocol. It defines a format for delivering audio and video files over the Internet.

Routing

Determining the path that a message or call has to take over circuits. TalkSwitch routes incoming calls according to the configuration.

Serial communication

A type of electronic communication requiring data bits to be sent one after the other, unlike parallel communication. Modems rely on serial communication to send data over telephone lines.

Shortcut key

A key combination that causes a specific command to be executed. Shortcut keys combine the Ctrl or Alt keys with other keys. (e.g. Ctrl and C is the shortcut key to copy selected items).

SIP

Session Initiation Protocol. A signaling protocol, developed by the Internet engineering task force (IETF).

SIP server

A server using the SIP protocol to manage calls between SIP devices.

Serial port

Serial ports are physical plug-ins. They allow two devices to exchange data.

SMTP

Simple Mail Transfer Protocol. The standard for e-mail transmission across the Internet. SMTP is a text-based protocol.

Terminal adapter

A device that allows analog devices to be used on digital lines (ISDN). Terminal adapters are similar to modems.

TX/RX

Transmit and receive data.

UDP

User Datagram Protocol. An Internet protocol which allows applications on a computer network to exchange short messages (datagrams).

uPnP

Universal Plug and Play. Allows networked devices to connect seamlessly in home and corporate environments using Internet-based communication standards.

USB

Universal Serial Bus. USB connection supports more bandwidth than a serial or parallel connection.

Virtual

The term 'virtual' refers to something that is not physically plugged in.

Voicemail (call answer)

A service provided by TalkSwitch or telephone companies giving callers the opportunity to leave a message when your phone is busy or unanswered.

VoIP

Voice over IP. Sending voice over an Internet connection.

VPN

Virtual Private Network. A communication network used within a company or organization to communicate over a public network. VPN message traffic is carried on Internet infrastructure using standard protocols.

INDEX

Numerics

3-way calling wait time list 199
3-Way Calling/Conference check box 134
3-Way Calling/Conference service 161, 185

A

About
  Call cascade 65
About page 25
Access code
  Adding 128
  Removing 128
Access code box 128
Access Code Configuration area 127
Account name box 167
Account Options area 169, 172
Action list 154
Action Performed After Auto Attendant Playback area 156
Actions During Auto Attendant Playback area 154
Activate Access Codes for Outgoing Calls area 127
Activate Access Codes for Outgoing Calls check box 127
Activate Account area 166
Activate Account check box 166
Activate Auto Attendant area 154
Activate Auto Attendant check box 154
Activate Auto Call Back area 169
Activate Auto Call Back check box 169
Activate Automatic Route Selection area 162
Activate Automatic Route Selection check box 162
Activate Extension area 58, 117
Activate Extension check box 58, 117
Activate Hunt Group area 150
Activate Hunt Group check box 150
Activate Line area 133
Activate line check box 133
Activate Profile area 48
Activate Profile check box 51, 52
Activate Prompted Call Back area 172
Activate Prompted Call Back check box 172
Activate Ring Group area 121
Activate Ring Group check box 121
Activate Speed Dial area 129
Activate Speed Dial check box 129
Activate Voicemail check box 75
Activate VoIP Number area 142
Activate VoIP Number check box 142
Active mailboxes only option 125
Add/Remove Lines button 151
Add/Remove Telephone Lines window 151
Adding
  Access code 128
  Counterpath IP phone 111
  Grandstream IP phone 106
  IP phones 81
  Polycom IP phones 102

TalkSwitch TS-350i and TS-550i IP phones 85
TalkSwitch TS-850i IP phones 96
TalkSwitch TS-9112i, TS-9133i and TS-480i IP phones 90
Additional Settings button 62, 63, 64, 65
Additional Settings window 62, 63, 64, 65
Adjust button 31
Administration area 27
All mailboxes option 125
Allow access to hunt groups check box 170, 173
Allow screening of calls option 185
Allow the use of same line connect (by dialing 80) check box 64, 167, 170, 173
Alternate DNS server box 34, 233
Analog extension 57
Announcements
recording 221
Answered call cascade 70
Answered tab 70
ARS Hunt Groups button 164
ARS Hunt Groups window 164
Attendant Console 56
Attendant Greetings area 157
Audio Control Options window 80, 183
Authentication area 49
Auto attendant
Receiving calls with 202
Receiving calls without 202
Troubleshooting 256
Auto attendant label box 154
Auto attendant message 156
Erasing 156
Loading 156
Recording 156
Auto Attendant Transfer Prompts area 187
Auto Attendants (Menus) page 154
Auto Attendants window 191
Auto call back
Recording announced message 170
Auto Call Back page 167, 168
Auto Route Selection area 163
Auto Route Selection page 162
Automatic Firmware Update window 180
Automatic route selection
Bypassing 164
Setting up 162
Automatically adjust clock for daylight saving changes check box 32
Automatically detect and switch to G.711 for fax over IP check box 194
Auto-Reboot check box 182

B

Browse button 39, 76, 157
Busy call cascade 67
Busy tab 67
Bypassing
Automatic route selection 164
Toll restriction 164

C

Calibrate Lines button 138, 195
Calibrate Lines window 137
Calibrating

Telephone lines 137
Call Back
Recording an announcement 170
Call Back Bridging Access window 170, 173
Call Back Options window 190
Call back ring return area 190
Call Bridge (DISA) page 166
Call cascade
About 65
Ignoring for calls transferred from local extensions 74
Ignoring for unanswered intercom calls 73
Call conference 208
SIP extension 213
Call detail record (CDR) 226
enabling 226
retrieving data 227
Call forward 217
screening options 218
to remote extension (music on hold) 222
Call handling
local extension 204
Call Handling area 65, 136, 143
Call Logging window 226
Call queue
cycling through queued calls 264
queuing and retrieving 206
to an extension ring group 207
Call Reminder Timers window 188
Call Reporting 56
Call routing
troubleshooting 257
Call transfer
from an analog extension 205
screened transfer 205
troubleshooting 257
unscreened transfer 205
Call Waiting check box 134
Caller ID / Time Set Options window 189
Caller ID Based Routing page 145
Caller ID check box 134
Caller ID lookup list area 146
Caller ID Name Tagging area 132
Caller ID Routing Groups area 146
Caller ID Settings area 65
Caller ID type to match list 146
Caller ID used for server provider list 65
Caller Options area 76
Carrier code 1 box 163
Carrier code 2 check box 163
Carrier Code Prefix area 163
Carrier codes 161
Case ID box 201
Cell/Pager Notification tab 78
Change Mode button 30
Change Mode window 30
Change Proxy button 35
Change Proxy window 35
Changing
mode 30
Check manually for update button 180
City or area code box 142

Cleaning XVII
Clear All Extensions button 128
Clear Code button 128
Clear telephone line after call transferred check box 185
CLID matching list area 146
CLID Routing Groups area 146
CLID type to match lisst 146
Clock setting area 189
CNG tones 194
Code name box 128
Codec options
  Setting 55
Codec Options button 55
Codec Options window 55
Configuration File window 24, 175
Configuration Selection page 23
Configuration software
  troubleshooting 255
Configure Auto-Detected TalkSwitch System button 23, 177
Configure Keys button 87, 88, 93, 94
Configure Notification Options window 79
Configure Notification Options... button 79
Configuring
  external IP extension 89, 98
  IP extension 96
  Router 235
Connect to a Different System button 23, 175
Connect to list 28
Connect to the following resource list 63
Connect using list 117, 130
Connect Via IP window 23, 175, 177
Connecting
  Extensions 11
  Fax machine 11, 14
  LAN 10
  Local area network 10
  Local extensions 11
  modem 13
  PA system 15
  Phones 11
  Single-line phone 12
  TalkSwitch unit to a PBX 21
  TalkSwitch unit to PC 6
  Telephone lines 11
  Two-line phone 12
Continuously run option 31
Counterpath 111
Country code box 142
Create a New Template File window 174
Create a Template File button 174
Current mode link 25
Current public IP address box 35

D

Date & Time area 32
Days to send area 227
Dedicated email address for sending/receiving emails box 43
Default gateway box 34, 233
Default language for system prompts to callers list 28
Delete any unrecognized inbound emails check box 45
Delete name from dial-by-name directory check box 126

Delete Password button 126
Delete password for mailbox list 126
Delete Password window 126
Delete Wav File button 40
Deleting
  Music on hold file 40
  Password 126
Detect 1 tone option 194
Detect 2 tones option 194
Diagnostics 2
Dial 0 routing area 28
Dial 9 routing area 28
Dial notification phone number using list 79
Dial-by-name directory
  Setting up 158
Digit Detection area 191
Direct line access 161
Direct Line Access area 61
Direct Line Access tab 62
Disable playing of transfer prompts to callers when redirecting from the auto attendant check box 187
Disable public IP address substitution check box 52
Distinctive ring 1 check box 134
Distinctive ring 2 check box 134
Do not disturb cascade 72
Do not disturb mode
  setting 73
Do Not Disturb tab 72

E

Edit button 170, 173
Electrical shock XVI
Email Address box 227
Email address box 41
Email Addresses window 77
Email log file area 227
E-mail notification 77
Email Notification Settings area 41
Email Notification tab 77
Email request interval list 45
E-mail server settings
  Testing 46
Email Server Settings area 43
Email Service page 40
Emergency service numbers XVI
Emergency Service Numbers area 161, 163
Enable automatic update of firmware via TFTP check box 180
Enable delete/save messages option check box 43
Enable email notification check box 77
Enable holiday mode check box 29
Enable NAT keep alives check box 52
Enable pager notification check box 79
Enable remote phone notification check box 78
Enabled check box 227
Ethernet connection 10
Exchange 47
Export list 132
Export System Speed Dial List area 131
Export System Speed Dial List window 132
Extension connectors 5
Extension ring groups

queuing callers to... 207
Extension tab 57
Extension type list 58
Extensions
Connecting 11
Numbering convention 17
Extensions Requiring an Access Code area 128
External audio source isolation XVII
External IP extension 57
configuring 89, 98
External IP extension calls list 50
External IP extension registration interval list 193
External IP extensions 82, 237
External Registration Interval area 193

F

Fax detection area 194
Fax Detection window 194
Fax machine
Connecting 11, 14
troubleshooting 258
Fax over IP area 194
File menu 174
File window 176
Firmware
Upgrading 6
Firmware detected 182
Firmware file 182
Firmware update files
Posting on your server 179
First name box 58, 117, 129
FQDN 35
From Auto Attendant tab 186
From Home Phone tab 187
Front panel 2
Full name box 41
Fully qualified domain name box 35
Function keys 87, 93
Programming TS-350i 87
Programming TS-9133i 93
Function list 87, 88, 93, 94

G

G.711 194
G.711 A-law check box 55
G.711 u-law check box 55
G.726 check box 55
G.729 check box 55
General Voice Mailboxes page 123
Global Message Waiting Indicator 2
Global Message Waiting Indicator area 124
Grandstream 106
Group label box 147

H

Handset ID 61, 98
Help and troubleshooting 255–261
Auto attendant 256
call routing/transfer/intercom 257
configuration 255

contacting TalkSwitch 261
fax machines 258
local area network 259
local extensions 258
music on hold 257
to obtain warranty service 261
Hold ringback reminder timer list 188
Holiday Mode label box 29
Holiday Settings window 30
Hotline Access area 63
Hotline Access tab 63
HTTP port box 193
HTTP Ports area 193
Hunt Group 80 is used for same line connect check box 164
Hunt Group Access area 167
Hunt Group Access window 167
Hunt group busy overflow for outgoing calls area 151
Hunt group check box 164
Hunt Group Line Assignments area 150
Hunt group name box 150
Hunt Group Permissions area 64
Hunt lines in the following order list 151
Hunt/Rollover/Busy Forwarding check box 134
Hunting order for outgoing calls area 151

I

Identify button 26
If a call is rejected from this extension list 70
If a fax call is detected list 155
If all lines are busy in this hunt group list 152
If being used with another PBX area 186
If busy or not answered after list 68
If Do Not Disturb is on for this ext. list 72
If extension is busy list 67
If the Extension Does Not Answer area 184, 185, 186
If the Extension is Busy area 184, 185, 186
If this extension is not answered after list 69
Ignoring
Call cascades for calls transferred from local extensions 74
Call cascades for unanswered intercom calls 73
Import an Email List area 43
Import an Email List window 43
Import System Speed Dial List area 131
Import System Speed Dial List window 131
Include ring group number in Caller ID relayed to extensions check box 122
Incoming lines
call detail record 228
using phones connected in parallel 216
Incoming mail server (POP3) box 44, 45
Incoming Mail Server Options area 45
Incoming Server Authentication area 45
Increasing
Voicemail memory 19
Installing
Memory card 19
TalkSwitch management software 5
TS-850iBase 96
Intercom
troubleshooting 257
Intercom call 189
Intercom Calls area 190

Intercom calls area 190
Intercom calls follow cascade settings check box 190
Intercom Calls Options window 190
Inter-digit timeout (IDT) list 163
Internal IP extension 57
Internal IP extension registration interval list 193
Internal Registration Interval area 193
Internet connection 11
Internet Email Settings window 43
Internet email settings window 44
Interval between attempts list 79
IP Configuration page 34
IP extension 57
  configuring 96
IP Extension Details area 59
IP Extension VoIP Lines area 193
IP Extensions window 193
IP phones
  Adding 81
Isolation XVII

K

Keep Alive Settings window 52
Key assignment template
  Saving 87
  Using 88
Keys 87, 93

L

Label box 169
LAN
  Connecting 10
  Optimizing TalkSwitch system 18
LAN connector 5
Language area 28
Language File Management window 28
Language of system prompts list 155
Last name box 58, 117, 129
Leading digits 160
Leading digits box 162
Licensed Add-Ons page 56
Lights 2
Line appearance 87, 93
Line connectors 5
Line disconnect time list 199
Line hunt group
  Restricting access by local extensions 164
Line Hunt Groups page 149
Line Optimization area 137
Line Reservation area 50
Line Reversal check box 135
Line Reversal Settings button 135
Line Reversal Settings window 135
Line type list 150
LineCalibration
  OPTIMIZEDMethod 137, 139
  QUICKMethod 137, 138
Load Greeting button 76, 157
Load Greeting window 76
Load Wav File button 39

Loading
Auto attendant message
Erasing 156
Local area network
Connecting 10
Networking TalkSwitch units 16
Optimizing TalkSwitch system 18
troubleshooting 259
Local extension
Bypassing automatic route selection 164
Bypassing toll restriction 164
Restricting access to line hunt groups 164
Local Extension/Fax page 57
Local extensions
Connecting 11
troubleshooting 257, 258
Location list 59
Location window 176
Lookup name directory
Setting up 158

M

MAC Address 86, 97
MAC address 61
MAC Selection window 60, 61, 86, 92, 97
Mailbox data
Viewing 125
Mailbox Greeting area 76
Mailbox Setting area 124
Mailboxes
Resetting 125
Main number box 134
Manual Port Mapping window 36, 83, 236
Maximum length box 192
Maximum recording length for each message list 124
Memory capacity
Increasing 19
Memory card
Installing 19
Memory cards 19
Memory slot 5
Memory usage
Viewing 178
Memory Usage window 178
Message length
Viewing 178
Message Waiting Indication window 80
Message waiting light 79
Message Waiting Light tab 80
Microsoft Exchange 47
Minimum length list 192
Minimum required time between DTMF digits list 191, 198
Minimum required time to detect a DTMF digit list 191, 198
Mode
changing 30
Mode 1 label box 29
Mode 1 tab 67, 69, 70, 72
Mode 2 label box 29
Mode 2 tab 67, 69, 70, 72
Modem
Connecting 13
Modes area 29, 30

More Settings button 43, 44
Move Down button 151
Move Up button 151
MUSIC connector 5
Music on hold
  Deleting file 40
  Playing on the PA system 40
  Setting up 15
  troubleshooting 257
My Outgoing Server (SMTP) Requires Authentication area 44
My Outgoing Server (SMTP) Requires Authentication check box 44
My TalkSwitch system link 26

N

Networking
  TalkSwitch units 16
No answer call cascade 69
No Answer tab 69
Notification option list 41
Notification Settings area 77
Notification Settings button 78
Notification Settings window 78
Notify the pager/remote phone after receiving list 124
NTP server box 33
NTP Server tab 33
Number box 142
Number of attempts list 79
Number of rings between attempts list 79
Number to call back box 169, 172
Number to trigger call back list 169
Numbering convention of extensions 17
Numeric message box 79

O

Obtain system time from caller ID check box 189
On-hold Settings area 39
On-Hold/Ringback page 38
Only remove messages shorter than list 125
Open a Configuration File or Template button 24, 174, 175
Open a File button 24, 175
Open File or Template window 24
Open Template button 88
Open Template window 88
OPTIMIZEDMethod
  LineCalibration 137, 139
Optimizing
  TalkSwitch system for network 18
Options menu 183
Other TalkSwitch button 177
Outbound proxy box 53
Outgoing mail server (SMTP) box 43, 45
Overflow tone notification area 152

P

PA connector 5
PA jack
  Voicemail screening 80
PA output option list 80, 183
PA system
  Connecting 15

Playing music on hold 40
Pager notification 79
Pager number box 79
Park ringback reminder timer list 188
Password
Deleting 126
Password box 44, 45, 49, 61, 143, 167
PBX
Connecting to a TalkSwitch unit 21
PBX extension length
Setting 159
PBX extension length list 186
PBX fraud XVI
PC
Connecting to a TalkSwitch unit 6
Perform a blind transfer option 185
Perform Busy at extension action option 184, 185, 186
Perform message notification for list 79
Perform No Answer at extension action option 184, 185, 186
Permissions page 127
Phone Line Services area 134
Phone notification 78
Phone Number area 142
Phone number box 78, 129, 142
Phone number used for TalkSwitch VoIP caller ID list 65
Phone Numbers area 134
Phones
Connecting 11
Play announcement only check box 76
Play notification tone when overflowing to another hunt group check box 152
Playback volume for music file list 39
Playing
Music on hold on the PA system 40
Polycom 102
POP3 service with Microsoft Exchange
Setting up 47
Populate calendar with holidays observed in list 30
Port box 23
Power and lightning surge protection XVI
Power connector 5
Power failure XVI
Preferred codec list 55
Preferred DNS server box 34, 233
Profile name box 48, 52
Programmable function keys 87, 93
Programming
TS-350i function keys 87
TS-480i softkeys 94
TS-550i softkeys 88
TS-9133i function keys 93
Prompted Call Back 171
Prompted call back
Recording announced message 173
Prompted Call Back page 171
Proxy server name box 53
Proxy/registrar server name box 49, 241
Public IP address area 35
Public IP address-checker server name box 35

Q

queue at extension 206
Queue options reminder timer list 188

QUICK calibration window 138
QUICKkMethod
  LineCalibration 138
QUICKMethod
  LineCalibration 137

R

Realm/domain box 53
Reboot button 183
Reboot System window 183
Recommended system specifications 1
Recording
  Auto attendant message 156
  Auto call back announced message 170
  Prompted call back announced message 173
Region Selection area 27
Register with authentication username 52
Registrar authentication list 49
Registrar entries
  Viewing 53
Registrar Entries window 53
Registrar server name box 53
Registration status
  Viewing 54
Registration Status area 143
Registration Status window 54
Regular Analog Extensions window 192
Remote Extension page 116
Remote phone number box 117
Remote Transfer area 185
Remove blank messages and hang up automatically check box 125
Removing
  Access code 128
Replace Caller ID name with box 146, 148
Re-registration interval list 188, 190
Reserve VoIP Lines button 50
Reset Mailboxes button 125, 126
Reset Mailboxes window 125
Resetting
  Mailboxes 125
Resource list 87, 88, 93, 94, 155
Restricting
  Local extension access to line hunt groups 164
Retry Auto Discovery button 177
Retry Auto-Discovery button 23
Ring back extension that transferred the call option 184, 185, 186
Ring Detection area 199
Ring group name box 121
Ring Group Options area 122
Ring Groups page 121
Ring Pattern Information window 199
Ring pattern list 122, 147
Ring Pattern Test button 199
Ring Pattern Test window 199
Ring Sequence window 136, 143, 147
Router
  Configuring 235
Router Port Forwarding area 36, 82, 235
Run Calibration button 138, 196

S

Save Template As button 87
Save Template As window 87
Saving
  Key assignment template 87
Scheduling area 31
Scheduling page 29
Select button 61
Select File button 43, 131
Select Wav File button 39, 157
Select Wav File window 39, 157
Selected Dates area 30
Server Port Numbers area 45
Service provider profile
  Setting up 50
Service provider VoIP network 244
Set Ring Group Extensions window 122
Setting
  Codec options 55
  do not disturb mode 73
  PBX extension length 159
  Unit ID 17
Setting up
  Automatic route selection 162
  Dial-by-name directory 158
  Music on hold 15
  POP3 service with Microsoft Exchange 47
  service provider profile 50
  TalkSwitch profile 48
  Toll restriction 162
Settings button 30, 52
Single Digit Action Timer area 191
Single digit fall through time list 191
Single-line phone 12
  Connecting 12
SIP server 231
SIP Server Registration area 188
SIP Server Registration Timer 200
SIP signalling port box 188
Slowly pulse the Power/Data light to indicate new messages check box 124
Softkeys 87, 93
  Programming TS-480i 94
  Programming TS-550i 88
Start time 1 list 163
Start time 2 list 163
Starting
  TalkSwitch management software 22
Starting RTP port box 189, 193
Starting SIP port box 193
Stop button 26
Subnet mask box 34, 233
Support window 201
Surge protection XVI
Switch modes depending on time option 31
Syslog server
  Setting up 181
System Information area 25
System IP Settings area 34
System name box 27
System password XVI
System password box 27

System prompt language for this extension list 117
System prompt language list 59
System specifications
  Recommended 1
System Speed Dials page 129
System time link 25

T

TalkSwitch 350i window 87
TalkSwitch 480i window 94
TalkSwitch 550i window 88
TalkSwitch 9133i window 93
TalkSwitch Command Console window 181
TalkSwitch configuration software
  Upgrading 6
TalkSwitch Date and Time Properties window 31
TalkSwitch Firmware Update window 182
TalkSwitch management software
  Installing 5
  Starting 22
  Upgrading 6
TalkSwitch memory cards 19
Talkswitch Network window 26
TalkSwitch Phone Programmable Key Functions 100, 214
TalkSwitch profile
  Setting up 48
TalkSwitch Proxy area 35
TalkSwitch systems
  Configuring 180
TalkSwitch TS-850i IP phones
  Adding 96
TalkSwitch unit
  Back panel connectors 2
  Connecting to a PBX 21
  Connecting to a PC 6
  Front panel 2
  Front panel lights 2
  Setting unit ID 17
TalkSwitch Unit area 227
TalkSwitch units
  Networking 16
  Upgrading 20
TalkSwitch VoIP network 239
TalkSwitch VoIP network calls list 50
Telephone lines
  Calibrating 137
  Connecting 11
  Precautions 11
Telephone Lines - Audio window 195
Telephone Lines - Detection and Timers window 198
Telephone Lines page 133
Telephones
  Single-line phone 12
  Two-line phone 12
Test Account Settings button 46
Test Account Settings window 46
Test email address box 46
Test Firewall window 37, 83, 200, 236
Test Ports button 37, 83, 200, 236
Test Ring Pattern button 199
Testing

E-mail server settings 46
TFTP server
Setting up 181
TFTP server location box 180
This server requires logon to incoming mail server before sending mail option 44
This server requires SMTP logon information option 44
This TalkSwitch location is the proxy/registrar checkbox 49
Time to send box 227
Time Zone area 32
Time zone list 60
To Local Extension tab 184
To Remote Extension tab 184
To Ring Group tab 186
Toll restriction 160
Bypassing 164
Setting up 162
Transfer and Clear area 185
Transfer and Clear check box 134
Transfer and Clear service 185
Transfer Preferences window 184
Transfer Prefix area 187
Transfer Settings area 40
Transmitted flash length list 199
TS-350i
Programming function keys 87
TS-480i
Programming softkeys 94
TS-50e expansion module 88
TS-550i
Programming softkeys 88
TS-850iBase
Installing 96
TS-9133i
Programming function keys 93
Two-line phone 12
Connecting 12
Type of phone list 58
Type of public address list 35

U

Unit ID
Setting 17
Unit IP address box 34, 233
Update button 182
Upgrading
Firmware 6
TalkSwitch configuration software 6
TalkSwitch management software 6
TalkSwitch units 20
UPnP 200
URL / IP address box 23
USB connection 10
USB connector 5
Use announced message check box 169, 172
Use automatic mode switching check box 31
Use call back prefix check box 169, 172
Use configured IP and DNS information option 34, 233
Use direct line access on check box 62
Use extension names in caller ID information for all outgoing VoIP calls option 49
Use password check box 169, 173
Use same line connect check box 118

Use system name in caller ID information for all outgoing VoIP calls option 49
Use the first and last names specified above to replace the incoming Caller ID names check box 133
Use the following call cascade settings for Holiday Mode list 30
Use transfer prefix list 187
User name box 44, 45
User/account box 143
User/account name box 49
Username and Password area 143
Username box 61
Using
  Key assignment template 88

V

Verifying the connections 21
View Blocked IP Addresses 200
View Mailbox Data button 125
View Registrar Entries button 53
View Registration Status button 54
Viewing
  Mailbox data 125
  memory usage 178
  Message length 178
  Registrar entries 53
  Registration status 54
Voice activity detection (VAD) check box 55
Voicemail
  E-mail notification 77
  Message waiting light 79
  Pager notification 79
  Phone notification 78
  Screening 80
Voicemail Management area 125
Voicemail memory
  Increasing 19
Voicemail Memory Usage window 26, 125
Voicemail messages link 26
Voicemail Preferences page 124
Voicemail tab 75
VoIP
  Setting up a service provider profile 50
VoIP Caller ID area 49
VoIP Configuration page 48, 50
VoIP FAQs 250
VoIP information 231
VoIP Network Administration Form 254
VoIP Numbers page 141
VoIP ports 200
VoIP ports area 188
VoIP Resource Reservation window 50
VoIP service provider calls list 50
VoIP Settings window 188

W

When a call is answered list 70
When listening to voicemail messages, always start with list 124
When matched, perform the following list 146